THE BASICS OF RADIO SYSTEM DESIGN

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Abstract
This paper is intended to give an overview of the design of radio transceivers to the engineer new to the field. It is shown how the requirements of a radio are derived from link budget calcul ations and then the modulation scheme, transmitter and receiver cascades are designed to fulfil them.

Introduction

In its basic form, RF system design is about connecting modules such as amplifiers, oscillators and mixers together in order to form a functional receiver or transmitter chain. However, to design a radio from scratch, the designer must have knowledge about the components that work with the radio such as the user interface, baseband functionality, power supply and antenna. Even before this, it important to analyse the customer requirements and radio link budget in order to produce a requirement specification.

In this paper, methods are introduced to calculate the required radio performance and then work through the process of cascading modules in order to form an RF or IF chain. This begins with gain calculations and then introduces gain compression, intermodulation, filtering and noise figure. In order to complete the overview of system design, the topics of modulation techniques and link budget trade-offs are briefly mentioned.

Link Budget

To transfer a radio signal from point A to point B, it is necessary to transmit a signal of adequate power in the right direction. The power of a radio signal of wavelength λ at a distance R in free space is described by (1).

$$Path\ loss = 20\log\left(\frac{4\pi R}{\lambda}\right) \qquad \text{dB} \tag{1}$$

In addition to free space path loss, the signal is also attenuated by obstructions such as vegetation, buildings and hills. It should be noted that free space path loss increases with frequency and so higher frequency bands are generally used for shorter range or line-of-sight communications. A number of path loss prediction models have been published which take the local radio environment into account. Models such as Hata [1] calculate a mean path loss for a large area whereas models such as Lee [2] separate out effects caused by natural and manmade structures to provide a detailed area specific prediction. In recent years, computer prediction models have become available which combine a path loss model with detailed mapping data, enabling the coverage of a transmitter in a particular location to be predicted.

In order to calculate a radio link budget, the power of the transmitter and the receiver's sensitivity are added to their antenna gains (if any) and to the path loss. For example, consider a transmitter of power 10dBm transmitting through a directional antenna of gain 7dBi to a receiver with 0dBi antenna and sensitivity of -110dBm. The maximum path loss that this link can suffer is the sum of the antenna gains and the difference between the transmit power and receive sensitivity, in this case 127dB. From (1), it can be calculated that in free space, this would be equivalent to a range of 35km at a frequency of 1GHz. However for a typical mobile radio environment, the range would be greatly reduced.

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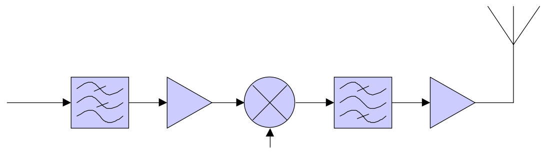
In addition to path loss, factors such as reflection, diffraction and interference affect the signal quality. There are a number of tools at the system designer's disposal to combat such degradations. The antenna is a particularly important factor as its height can be increased to 'see' over obstacles. A high gain antenna can be employed to combat path loss in a certain direction or to avoid receiving interference from another direction. Two or more antennas can be used in a 'diversity' scheme where the receiver (or transmitter) chooses to use the antenna with the best quality signal at a particular moment in time.

Digital techniques are also increasingly being used to combat interference and reduce errors in digital systems. Digital equalisers are used to compensate for channel imperfections such as reflected versions of a signal which arrive shortly after the direct signal (similar to picture ghosting on analogue terrestrial television). Forward error correction digitally encodes the data to be sent to enable errors to be detected and corrected in the receiver. It is used in many applications such as microwave point to point links, mobile telephony and telemetry.

Once a link budget has been calculated and suitable component technologies have been chosen, a requirements specification can be written, from which the cascade of radio modules can be designed.

Gain Cascade

In radio, it is easiest to work with gain described in decibels and power in dBm (decibel relative to 1mW) as these can simply be added together when connecting modules in a chain. An example of a transmitter gain line-up is shown in Figure 1.



	Filter	Amp	Mixer	Filter	Amp
Level In/dBm	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]
Level Out/dBm	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]
Gain/dB	-1	15	-8	-2	25

Figure 1 Transmitter gain line-up

1dB Compression Point

When working with a gain cascade, it is necessary to consider the level of signal being applied to each module because components such as mixers and amplifiers can only output a finite amount of signal power. A useful measure of the amount of power that a device can produce is the '1dB compression point'. At low signal levels, a device is considered linear but as the input signal is increased, the amount of signal at the output will begin to tail off. When the difference between the input and output reaches 1dB, the input power or output power are measured and referred to as the 'input 1dB point' or 'output 1dB point'.

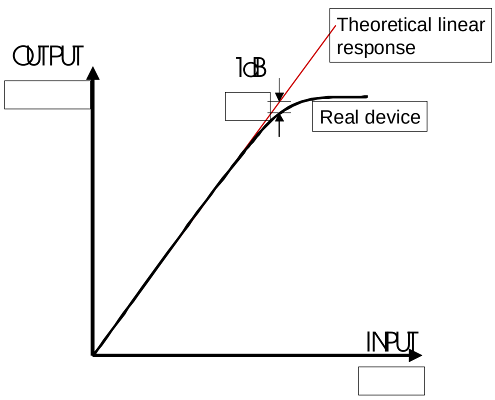


Figure 2 1dB Compression Point

For a transmit or receiver chain, the level of signal at components such as amplifiers and mixers should be compared with their 1dB compression points to ensure that they are not exceeded. Even components commonly thought of as linear such as filters should also be checked to ensure that their maximum rated power is not exceeded.

Third Order Intercept Point

The third order intercept point is a measure of linearity which describes the amount of third order harmonic that can be expected in a device. As a signal's amplitude is clipped by devices such as a mixers and amplifiers, harmonics are produced. Third order products are important as unlike second order products, they fall near to the wanted frequency. In order to measure a device's third order harmonics, two tones f_1 and f_2 are applied to the input and the third order harmonics can be viewed on a spectrum analyser as shown in Figure 3.

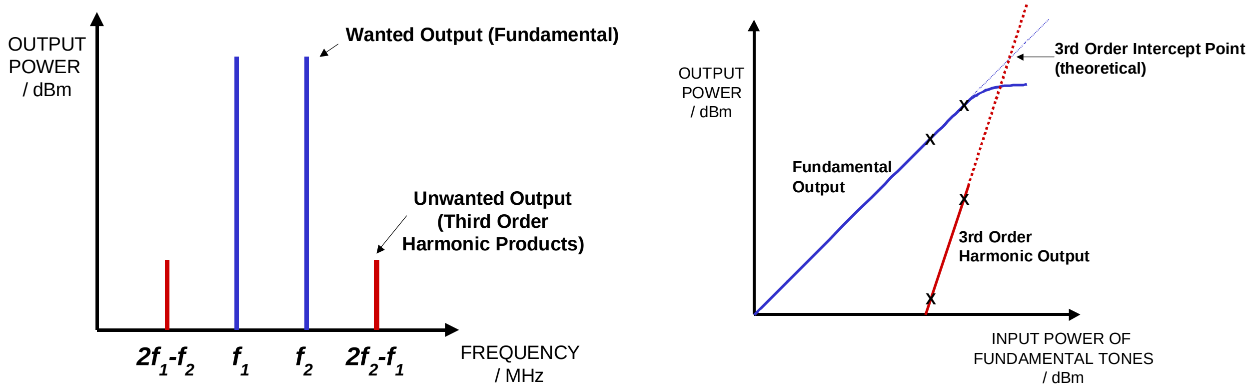
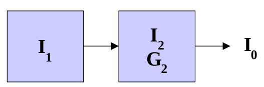


Figure 3 Third Order Intercept Point

The level of fundamental and third order output are plotted and the two lines are extended to the theoretical point where they would cross. It should be noted that the third order intercept point (IP3) is theoretical and will change slightly depending on the signal level from which it is extrapolated.

For a cascade of RF modules, it is often useful to show the cascaded IP3. The expression for the IP3 of two cascaded blocks is described in equation (2).



$$I_0 := 10\cdot\log\left(\frac{1}{\frac{1}{10^{I_2}}+\frac{1}{10^{I_1+G_2}}}\right) \qquad \text{All terms in dB} \tag{2}$$

The IP3 can be referred to the input or output of a module and care should be taken to always specify which is meant in order to avoid confusion. This is especially important in a transmit or receive cascade because all the modules must be described in the same way in order for their cascaded IP3 to be calculated.

Filtering

Consider the receiver shown in Figure 4. The first downconversion from RF to IF involves mixing the RF input with the local oscillator and the IF signal will consist of LO+RF and also LO-RF. For example, RF input at both 1.9GHz and 2.1GHz would mix with a 2GHz LO into an IF signal of 100MHz. An 'image filter' is used in order to select the required RF input band and attenuate the unwanted image response. The bandpass function of the image filter can also be designed to attenuate other frequency bands which might otherwise be received and overload the input amplifier.

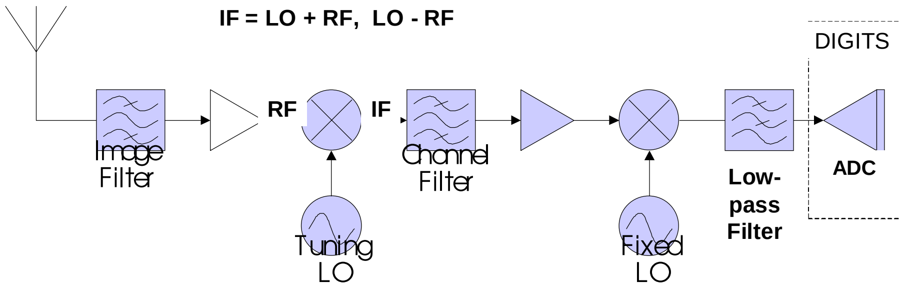


Figure 4 Filtering in a Receiver

The IF signal at the output of the first downconversion contains a whole band of signals. A narrowband filter is used to select the wanted channel and attenuate other channels in the band. As the centre frequency of the narrowband filter is fixed, the tunable local oscillator is used to tune the wanted channel to the centre of the filter passband. Alternatively, the first mixer can be used to upconvert the RF to a higher IF. This is used where the RF bandwidth is high with respect to its centre frequency because the higher IF allows the tunable oscillator to have a proportionally lower tuning range.

The final mixer converts the IF down to baseband, ready for input into the demodulator. A low pass filter is used to reduce high frequency noise into the analogue to digital converter and to prevent aliasing of the wanted signal.

A wide variety of filtering schemes are used in transmitter and receiver chains, but the concept of RF bandpass filter, channel selection filter and low pass baseband filter can be applied to many designs.

Thermal Noise

The noise power from a resistor or resistive source is given by equation (3) and is used to define the thermal noise floor in a given bandwidth. A useful number to remember is that at a temperature of 290K (room temperature), the thermal noise floor is -204dBW or -174dBm in a 1Hz bandwidth.

$$N = 10\log(KTB) \text{ dBW} \qquad (3)$$

where $K = 1.38\times10^{-23}$ Boltzmann's Constant
$T =$ temperature in kelvin
$B =$ bandwidth in hertz

Noise Figure

The noise figure is a measure of the amount by which a device increases the noise power, usually assuming that the input noise temperature is 290K. Figure 5 illustrates an example.

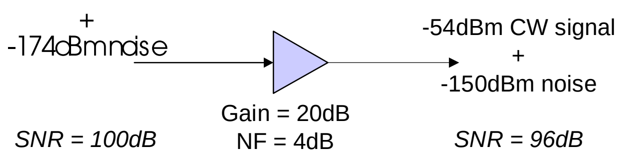
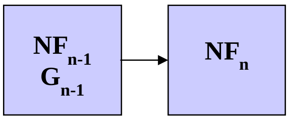


Figure 5 Noise Figure

An alternative method of expressing the noise introduced by a device is to give its equivalent noise temperature (3). Equation (4) describes the cumulative noise figure of two cascaded modules. It can be used for modules such as filters and passive mixers which do not have gain if their loss in decibels is used in place of their noise figure.



$$Cumulative\ NF_n = 10\log\left(10^{\frac{NF_{n-1}}{10}} + \frac{10^{\frac{NF_n}{10}} - 1}{10^{\frac{G_{n-1}}{10}}}\right) \qquad (4)$$

Figure 6 illustrates the importance of the front end noise figure and gain on the overall cumulative noise figure of a receiver. The first case shows a low noise amplifier before a bandpass image filter. This gives a low cumulative noise figure of 2.2dB at the output of the mixer but the low noise amplifier is left unprotected from large out of band interfering signals. The second case is identical except that the positions of the filter and amplifier have been swapped. The cumulative noise figure has increased considerably but the front-end amplifier is now protected from unwanted out of band interferers.

	Amp	Filter	Amp	Mixer
Level In / dBm	-100	-85	-87	-70
Gain / dB	15	-2	17	-8
Cumulative Gain	15	13	30	22
NF	2	2	3	8
Cumulative NF	2.0	2.1	2.2	2.2
Level Out / dBm	-85	-87	-70	-78

	Filter	Amp	Amp	Mixer
Level In / dBm	-100	-102	-87	-70
Gain / dB	-2	15	17	-8
Cumulative Gain	-2	13	30	22
NF	2	2	3	8
Cumulative NF	2.0	4.0	4.1	7.1
Level Out / dBm	-102	-87	-70	-78

Figure 6 Cascading Noise Figures

Phase Noise

Phase noise from a local oscillator mixes with the wanted modulated signal during frequency mixing in the transmitter and receiver. One of the results is that in the frequency domain, in addition to its modulation sidebands, the wanted signal also has phase noise sidebands. The effect can be seen in the time domain as a phase error on the modulation. Poor phase noise performance can also give rise to undesirable levels of unwanted adjacent channel power being radiated by transmitters and poor adjacent channel selectivity in receivers. With knowledge of the transmitted spurious level requirements and demodulator characteristics, the system designer can specify the level of phase noise at various frequency offsets to the wanted tone, thus preventing excessive transmitted noise and demodulation errors.

Single Sideband Modulator

It is assumed that the reader is familiar with the multiplying act ion of a mixer, resulting in the production of two tones in the frequency domain as described in equation (5).

$$2\cos(A)\cos(B) \equiv \cos(A+B) + \cos(A-B) \qquad (5)$$

The action of a mixer results in two products, of which one is normally the wanted and the other one unwanted, necessitating the need to filter one of them out. The addition of an extra mixer and two 90 degree phase shifters as shown in Figure 7, allows one of the sidebands to be cancelled out (6).

$$\sin(A)\sin(B) + \cos(A)\cos(B) \equiv \cos(A-B) \qquad (6)$$

Equation (7) also shows that by simply swapping the oscillator phase shifter outputs, the other sideband can be selected.

$$\sin(A)\cos(B) + \cos(A)\sin(B) \equiv \sin(A+B) \qquad (7)$$

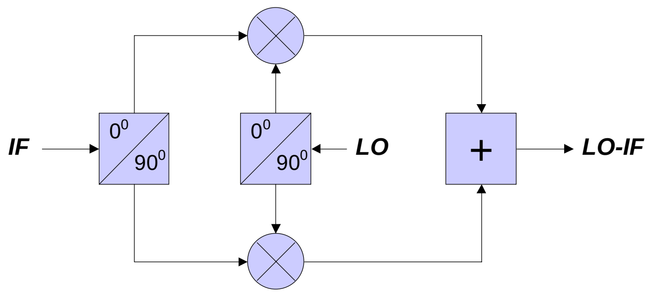


Figure 7 Single Sideband Mixer

This arrangement is typically used for

- Single sideband modulation (SSB). This is the same as standard telephony amplitude modulation but with one side of the modulation removed. The signal still contains the same information but requires less bandwidth.
- Image reject mixer. When the single sideband mixer is used in a receiver, it can be used to aid rejection of the unwanted image frequency in the downconversion. The same principle can be used in transmitters to reduce the need for filtering of the unwanted mixer product.

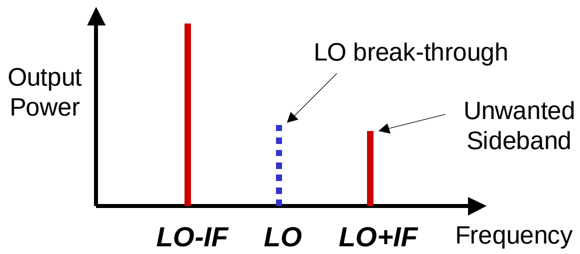


Figure 8 Single Sideband Modulator / Image Reject Mixer

Figure 8 illustrates the output of an SSB modulator. It can be seen that despite the mathematics of (6) and (7), the non-ideal multiplying of real-world mixers leads to some unwanted sideband and local oscillator at the output. This is usually due to small DC offsets on the inputs to the mixers which can be minimised by calibration using adjustable potential dividers. A typical SSB modulator can achieve 10 to 25dB of unwanted sideband rejection but with the aid of calibration, this might be increased to 40 or 50dB.

IQ Modulation

Figure 9 shows a modification to the SSB modulator. Data to be transmitted is separated into two streams with a serial to parallel converter. In the case of Quadrature Phase Shift Keying (QPSK) modulation, for each two bits of data, one is sent to the I and one to the Q channel. This creates a constellation of I and Q with four 'symbols', each of which is described by two bits.

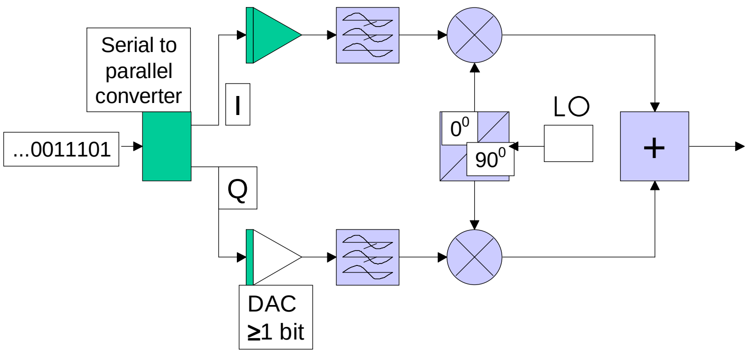


Figure 9 IQ Modulator

As the digital data has fast edges and contains harmonics, a low pass filter is used to shape the modulation and restrict its bandwidth. The filters used for QPSK are of a raised cosine type which shape the data to minimise interference between consecutive symbols, known as inter-symbol interference. Another common filter is the Gaussian filter used for the DECT and GSM communication standards. The filters can be either analogue or digital, with the digital to analogue converters in Figure 9 just before or after the filter blocks as appropriate.

To extend this principle further, the serial to parallel converter can output a word containing more than one bit on each of the I and Q channels. With two bits per I and Q channel, four bits describe each symbol. This is known as 16 QAM (Quadrature Amplitude Modulation) because there are 16 possible symbols and is illustrated in Figure 10.

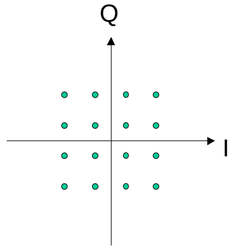


Figure 10 Constellation of 16 QAM

As one might expect, other modulation schemes 2^n QAM which contain *n* bits per symbol are possible. The advantage of each of these greater numbers of bits per symbol is that each uses the same amount of radio bandwidth for a given symbol rate. For example 256 QAM (8 bits per symbol) can carry 40Mbps in the same bandwidth that 16 QAM (4 bits per symbol) can carry 20Mbps because they both hav e a symbol rate of 5Msymbols/s. Proakis [3] gives a good technical description with calculations for error rates and required bandwidth. Alternatively, [4] is a descriptive overview and easier to read.

Unfortunately such advantages come at the cost of the amount of signal to noise ratio (SNR) required by the different modulation schemes in order to maintain the same bit error rate. This is why high order modulation schemes are often used for fast data rate point to point microwave links which can guarantee good SNR, whereas most mobile telephone systems use low order schemes with only two bits per symbol because communication range (coverage) is of utmost importance. Each radio system has its modulation scheme chosen depending upon the importance of bandwidth, transmission power available, range required, data rate and the complexity and cost of the required modulators and demodulators.

Conclusion

It can be seen that the process of designing a complete radio system begins with a study of the requirements to produce a link budget and performance specification. The specification is then used to design transmitter and receiver cascades which fulfil these requirements. The areas of propagation and modulation techniques are important to the designer for the definition of new radio systems and whilst they have been briefly introduced here, it is recommended that the reader follows references such as [3] and [4] to find out more.

The design of a transmitter or receiver is usually an iterative process of compromise between performance, size, cost and power consumption. The system designer needs to have a good overview of all these in order to produce sensible specifications!

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[3] PROAKIS, J:*'Digital Communications'*, Chapter 5, 3rd Edition, pub. McGraw-Hill 1995

[4] Hewlett Packard '*Digital Modulation in Communications Systems - An Introduction'*, Application Note 1298, 1997