Sample Solution

Case Study Competition 2008

"Get The Signals"

Question 1: State the differences between the characteristic features of a monitoring receiver and a spectrum analyzer. (10 points)

Spectrum analyzer

- No preselection
- Mixer directly at the input
- High measurement accuracy
- Operating concept optimized for measuring tasks
- No AGC loop
- Filter optimized for measuring tasks (CISPR)
- Calibration at regular intervals (overall calibration)
- Standard-oriented analysis and presentation of results
- Special marker function limit lines
- IEC/IEEE bus control
- Special functions are task-specific

Many of the functions that are essential for radio acquisition, such as FSCAN, MSCAN, panorama, squelch, dwell time, etc., are not necessary and are therefore not present or only present in a very limited form.

Monitoring receivers

- Built-in preselection
- AGC loop with optimized time constants
- Signal switching using semiconductors (no relays)
- Operating concept optimized for signal acquisition tasks
- Realtime-capable processor design
- Special functions (e.g. squelch)
- Functions such as FSCAN, MSCAN and panorama (RF and IF)
- Internally controlled processes (scans)
- Audio processing is essential
- Mains and battery-powered operation
- High reliability and MTBF (security-related)
- Comprehensive self-test concept – short MTTR
- Easy system integration
- General-purpose interfaces for supplementary equipment
- Low power consumption
- Elevated temperature range
- High mechanical robustness for mobile use
- Stringent EMC requirements

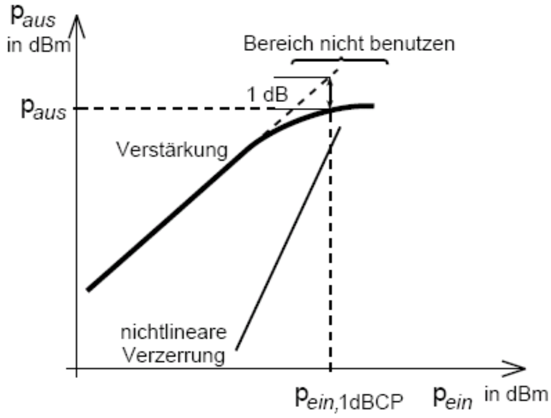
Question 1	Teamname				
	10 points				
	Teamname				
	10 points				

Question 2: State the definition of the 1 dB compression point and explain why IP1 is an important design parameter for receivers. (5 points)

Every active transfer block has a linearity limit above which the amplification factor is no longer constant. The amplification decreases gradually. For this reason, it is common practice to designate the point pin,1dBCP at which

pout – pin,1dBCP = v – 1 dB

as the 1 dB compression point. Here v represents the amplification factor (voltage gain) of the block, which is assumed to be constant.
If a transfer block is required to have high linearity, the input level must always be significantly lower than the 1 dB compression point pin,1dBCP. This requirement holds true independently of any additional nonlinear distortion that may occur. With a circuit consisting of a series of chained blocks, the input signal level relative to the 1 dB compression point must be checked for each block individually in order to avoid undesired first-order distortion products. It is not possible to determine a meaningful overall 1 dB compression point for a set of chained amplification stages.



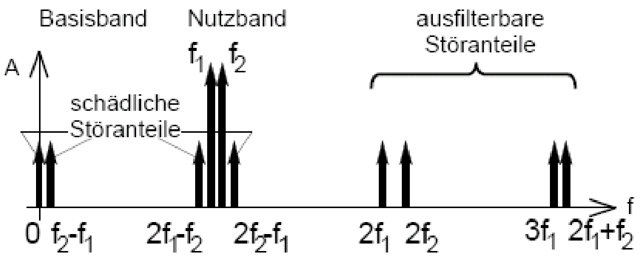
Bereich nicht benutzen	Do not use this region
Verstärkung	Gain
nichtlineare Verzerrung	Nonlinear distortion
Paus in dBm	pout in dBm
Pein in dBm	pin in dBm
Pein,1dBCP	pin,1dBCP

Question 2	Teamname				
	5 points				
	Teamname				
	5 points				

Question 3: State the definition of IP3. (5 points)

Nonlinear distortion at orders higher than the first order is characterized by specifying an intercept point for each transfer block. The intercept point can be referenced to the input or the output. Here
OIPn = IIPn + v
where OIPn is the nth-order intercept point referenced to the output, IIPn is the nth-order intercept point referenced to the input, and v is the gain of the linear stage. The third-order intercept point is obtained when n = 3.

Intermodulation arising from third-order nonlinearities produces (among other things) interference components at 2f1 – f2 and 2f2 – f1. These effects can be illustrated as shown in the following figures.



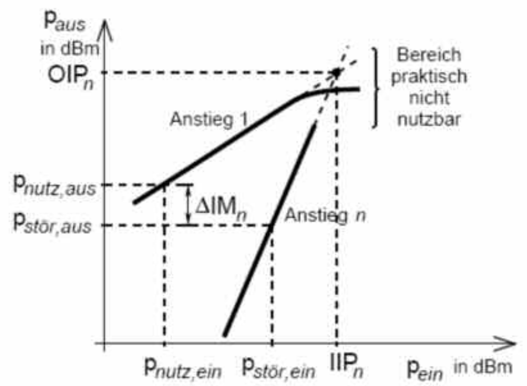


Illustration of the intercept point and intermodulation margin

Basisband	Baseband
schädliche Störanteile	Detrimental interference components
Nutzband	Signal band
ausfiltbare Störanteile	Filterable interference components
Pnutz, aus	psig, out
Pstör, aus	pinterf, out
Pnutz, ein	psig, in
Pstör, ein	pinterf, in
Pein	pin
Paus	pout
Bereich praktisch nicht nutzbar	Region not usable in practice
Anstieg	Slope

Question 3	Teamname				
	5 points				
	Teamname				
	5 points				

Question 4: Explain why the slope of the intermodulation line is 3. (10 points)

The input signal is given by

$$x = A\sin(y) + B\sin(z) \tag{2.62}$$

definiert, wobei A und B die Amplituden und $y = 2\pi f_1$ bzw. $z = 2\pi f_2$ die Kreisfrequenzen mit $f_1 \approx f_2$ darstellen. Als Beispielblock dient ein Verstärker mit der polynomial approximierten Kennlinie

$$f(x) = v_1 x + v_2 x^2 + v_3 x^3. \tag{2.63}$$

Dabei symbolisiert v_1 die Verstärkung des linearen Anteils. v_2 bzw. v_3 stellen die Faktoren des quadratischen bzw. kubischen Kennlinienanteils dar. Ausführlich kann man dann für das Verstärkerausgangssignal schreiben

$$\begin{aligned} f(x) = & + v_1 \overbrace{(A\sin(y) + B\sin(z))}^{\text{Nutzsignal}} \\ & + v_2 \overbrace{\left(\frac{1}{2}A^2 + AB\cos(y-z) + \frac{1}{2}B^2\right)}^{\text{Störanteil 2. Ordnung (Basisband)}} \\ & - v_2 \overbrace{\left(\frac{1}{2}A^2\cos(2y) + AB\cos(y+z) + \frac{1}{2}B^2\cos(2z)\right)}^{\text{Störanteil 2. Ordnung (doppelte Nutzbandfrequenz)}} \\ & + v_3 \overbrace{\left(\frac{3}{4}A^3\sin(y) + \frac{3}{2}AB^2\sin(y)\right)}^{\substack{\text{verfälschender Nutzsignalanteil I} \\ \text{von Nichtlinearität 3. Ordnung (Kreuzmodulation)}}} \\ & + v_3 \overbrace{\left(\frac{3}{2}A^2B\sin(z) + \frac{3}{4}B^3\sin(z)\right)}^{\substack{\text{verfälschender Nutzsignalanteil II} \\ \text{von Nichtlinearität 3. Ordnung (Kreuzmodulation)}}} \\ & + v_3 \overbrace{\left(\frac{3}{4}A^2B\sin(2y-z) - \frac{3}{4}AB^2\sin(y-2z)\right)}^{\text{Störanteil 3. Ordnung (Nutzband)}} \\ & - v_3 \overbrace{\left(\frac{3}{4}A^2B\sin(z+2y) + \frac{3}{4}AB^2\sin(y+2z)\right)}^{\text{Störanteil I 3. Ordnung (dreifache Nutzbandfrequenz)}} \\ & - v_3 \overbrace{\left(\frac{1}{4}A^3\sin(3y) + \frac{1}{4}B^3\sin(3z)\right)}^{\text{Störanteil II 3. Ordnung (dreifache Nutzbandfrequenz)}}. \end{aligned} \tag{2.64}$$

Definiert, wobei A und B die Amplituden und y = 2πf1 bzw. z = 2πf2 die Kreisfrequenzen mit f1 ≈ f2 darstellen. Als Beispielblock dient ein Verstärker mit der polynomial approximierten Kennlinie	**Where A and B are the amplitudes and y = 2πf1 and z = 2πf2 the angular frequencies of the signals f1 and f2, respectively, with f1 ≈ f2. As an example, consider an amplifier with a characteristic curve approximated by the polynomial expression**
Dabei symbolisiert v1 die Verstärkung des linearen Ateils. v2 bzw. v3 stellen die Faktoren des quadratischen bzw. kubischen Kennlinienanteils dar. Ausführlich kann man dann für das Verstärkerausgangssignal schreiben	**Here v1 is the amplification factor of the linear component, while v2 and v3 are the coefficients of the quadratic and cubic components of the characteristic curve. The output signal of the amplifier can be expressed in expanded form as follows:**
Störanteil 2. Ordnung (basisband)	**Second-order interference component (baseband)**
Nutzsignal	**Useful signal**
Störanteil 2. Ordnung (doppelte Nutzbandfrequenz)	**Second-order interference component (at twice the signal band frequency)**
verfälschender Nutzsignalanteil I von Nichtlinearität 3. Ordnung (Kreuzmodulation)	**Signal distortion component 1 from third-order nonlinearity (cross-modulation)**
verfälschendes Nutzsignalanteil II von Nichtlinearität 3.	**Signal distortion component 2 from third-order**

Ordnung (Kreuzmodulation) Störantiel 3. Ordnung (Nutzband)	nonlinearity (cross-modulation) Third-order interference component (signal band)
Störanteil I 3. Ordnung (dreifache Nutzbandfrequenz)	Third-order interference component 1 (at three times the signal band frequency)
Störanteil II 3. Ordnung (dreifache Nutzbandfrequenz)	Third-order interference component 2 (at three times the signal band frequency)

If both input signals have the same amplitude, it can be seen that in the case of the third-order intermodulation products that lie in the signal band, increasing the input signal level by 1 dB causes the output signal level to rise by 3 dB. This corresponds to a slope of 3 for the line.

Question 4	Teamname				
	10 points				
	Teamname				
	10 points				

Question 5: Calculate the overall IP3 for a chained pair of amplifiers (amplifier 1: gain G = 10 dB, IP3 = +5 dBm; amplifier 2: gain G = 8 dB, IP3 = +10 dBm). (10 points)

For a chained series of m two-port networks, the nth-order intercept point can be defined using the following formula:

$$IP_n^{(1-n)/2} = IP_{n,m}^{(1-n)/2} + (G_m IP_{n,m-1})^{(1-n)/2} + (G_m G_{m-1} IP_{n,m-2})^{(1-n)/2} + \ldots + (G_2 G_m IP_{n,1})^{(1-n)/2}$$

With the values stated for the question, the calculated value of the overall IP3 is 8.24 dBm.

Question 5	Teamname				
	10 points				
	Teamname				
	10 points				

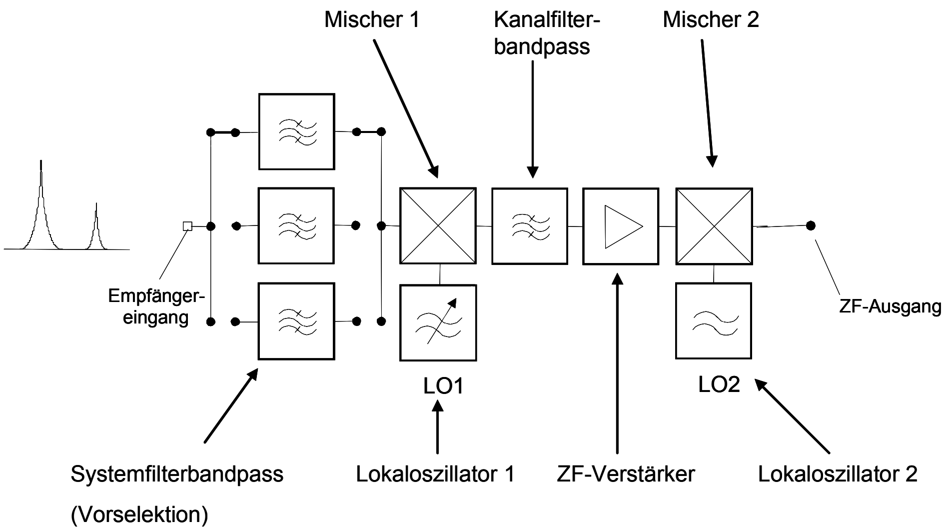
Question 6:

6.1: What sort of receiver is this? **(5 points)**

6.2: Explain the purpose of the individual functional blocks. **(10 points)**

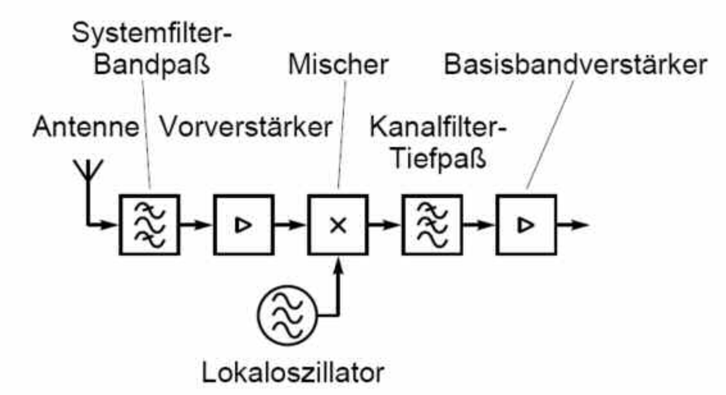
6.3: Describe the advantages and disadvantages of this design and compare it with a direct-conversion receiver. **(10 points)**

The illustrated receiver is a double-conversion superheterodyne receiver (also called a double superhet receiver). Here the signal first passes through a preselection filter, and the resulting reception band is converted to an intermediate frequency and then filtered again. The intermediate frequency is then processed further in the same manner as with a direct-conversion receiver. The amplification and selection functions are distributed over the individual stages and occur at different frequencies, so parasitic coupling between the stages does not cause any problems. The signal dynamic range in the stage corresponding to the direct-conversion receiver is usually less than 40 dB. The advantage of this arrangement is that the gain, linearity and selection requirements for the individual components are generally not as severe as with a direct-conversion receiver. The purposes of the individual functional blocks can be seen from the following figure.



Empfängereingang	**Receiver input**
ZF-Ausgang	**IF output**
Systemfilterbandpass	**System bandpass filter**
(Vorselektion)	**(preselection)**
Lokaloszillator 1	**Local oscillator 1**
Kanalfilterbandpass	**Channel bandpass filter**
Mischer	**Mixer**
ZF-Verstärker	**IF amplifier**

A disadvantage is that a superheterodyne receiver has a second frequency range in which it can receive signals. This range, which is called the image frequency range, must be very carefully suppressed. This is one of the main problems in designing a superhet receiver. In addition, it has significantly more modules than a direct-conversion receiver. Due to the need for a narrow transition between the passband and stopband regions, the intermediate frequency channel filter is usually implemented as a ceramic filter or surface acoustic wave filter in current designs, which means it resists all efforts to incorporate the filter in an integrated circuit. The local oscillator frequencies must be selected carefully in order to eliminate other spurious reception frequencies.



Depiction of a direct-conversion receiver (DCR)

Systemfilter-Bandpaß	System bandpass filter
Mischer	Mixer
Basisbandverstärker	Baseband amplifier
Antenne	Antenna
Vorverstärker	Preamplifier
Kanalfilter-Tiefpaß	Low-pass channel filter
Lokaloszillator	Local oscillator

An advantage of the direct-conversion principle is that it requires relatively few components. The channel low-pass filter can be implemented as an active filter due to the low baseband frequency, which means it can be fully implemented in an integrated circuit. Only one local oscillator frequency has to be generated. The difficulties associated with amplifying high-frequency signals, which requires a high current level to achieve adequate linearity, are largely avoided because most of the amplification occurs in the baseband.

Question 6	Teamname				
6.1	5 Points				
	Teamname				
6.1	5 points				

Question 6	Teamname				
6.2	10 points				
	Teamname				
6.2	10 points				

Question 6	Teamname				
6.3	10 points				
	Teamname				
6.3	10 points				

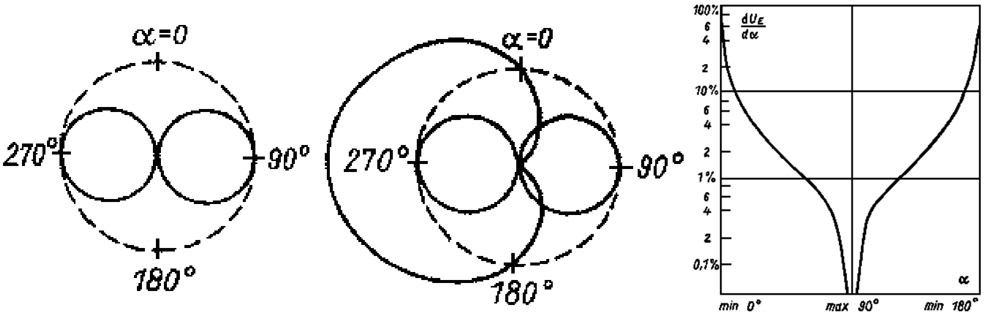
Question 7: Describe the effect of the indicated oscillator radiation on the reception performance of the receiver. (5 points)

A portion of the oscillator radiation indicated in the figure reaches the antenna as an oscillator signal. As the antenna has a finite matching impedance (reflection coefficient ≠ 0), this signal in turn arrives at the input of the first local oscillator. If two signals with the same frequency are multiplied, this produces a DC voltage that can drive the following stages into saturation. However, this can be avoided by using a consistently AC-coupled system design.

Question 7	Teamname				
	5 points				
	Teamname				
	5 points				

Question 8: Describe the principle of minimum-signal direction finding and explain its advantages and disadvantages relative to maximum-signal direction finding. (15 points)

If the angle between the direction to the transmitter and the orientation of the ferrite core changes, this yields the well-known figure-eight antenna pattern of the ferrite antenna or a cardioid pattern if an auxiliary antenna is also used.



As the human ear can only distinguish loudness differences of 1 dB (10 %) or more, the null point of the figure-eight antenna pattern is used for direction finding because the received voltage changes most rapidly around the null point when the receiver is rotated, as shown by the following figure. The null point of the cardioid pattern is not used for precise direction determination because it is much less pronounced and because it can be degraded very easily by changing conditions (electromagnetic field, distance from the transmitter, height of the receiver, changes in matching, etc.)

The achievable direction-finding accuracy depends on several factors:

- **The difference between the direction of propagation of the electromagnetic field at the receiver location and the direction to the transmitter**
- **Subjective DF operator errors**
- **The DF deviation, i.e. the fixed deviation between the orientation of the reception minimum and the direction of propagation of the electromagnetic field**
- **The DF resolution, i.e. the difference between the two directions at which a specific change in the received signal is observed (see Fig. 8)**

Advantages:
- **The minimum is significantly more pronounced than the maximum**
- **Very simple method**

Disadvantages:
- **The minimum is more difficult to hear**
- **The minimum is not unambiguous**
- **Cannot be used to determine the direction of short signals**

Question 8	Teamname				
	15 points				
	Teamname				
	15 points				

Question 9: What aids can be used to obtain an unambiguous DF result? (5 points)

One of the two minimums can be eliminated by evaluating the signal from a supplementary omnidirectional antenna.

Question 9	Teamname				
	10 points				
	Teamname				
	10 points				

Question 10: On the basis of the following figure, describe the working principle of the Watson–Watt arrangement. Describe the advantages and disadvantages of this arrangement. (10 points)

If the signals from the crossed loop antennas are fed to the inputs of an X-Y oscilloscope, the electron beam is rotated in proportion to the angle of the incident wave. As the electron beam follows this angle with practically zero inertia (compared with mechanical manipulation), the result is an extremely short response time. A Watson–Watt direction finder needs at least xxx (exactly matched) receiver channels. The deflection of the electron beam is proportional to the cosine of the angle of incidence in the vertical direction and the sine of the angle of incidence in the horizontal direction. The following figure illustrates how the image is generated.
The DF signal voltages, which have a sinusoidal shape as a function of time, generate a Lissajous figure, which in the general case is an ellipse. The following expressions for the two deflection voltages are only valid when the two DF signals are in phase or 180 degrees out of phase, which only occurs in an undisturbed field:

$U_y = A\, V_y \cos(p) \sin(\omega t)$
$U_x = A\, V_x \sin(p) \sin(\omega t)$

Here Uy corresponds to the north–south direction and Ux to the east–west direction. A represents the maximum voltage available to the direction finder and thus takes into account at least the field strength, the effective height of the antenna, and all losses up to the DF input. Vy and Vx are the amplification factors of the two channels, and p is the angle of incidence of the wave with reference to the north direction, as projected onto the plane. The wave is assumed to be a sinusoidal function of time with an angular frequency ω, and t is the time variable.
The ambiguity of the measurement can be eliminated by using an additional omnidirectional antenna (the negative halfwave is used for blanking, and the field strength of the incoming wave is converted into a proportional brightness of the oscilloscope beam).

Advantages of the arrangement:
- Very short signal durations can be captured
- The arrangement is easily implemented
- Compact construction is possible

Disadvantages:
- Systems with small apertures ($D/\lambda < 0.2$) are susceptible to multi-path propagation errors
- Large direction errors occur when space waves with high elevation angles are received

Question 10	Teamname				
	10 points				
	Teamname				
	10 points				