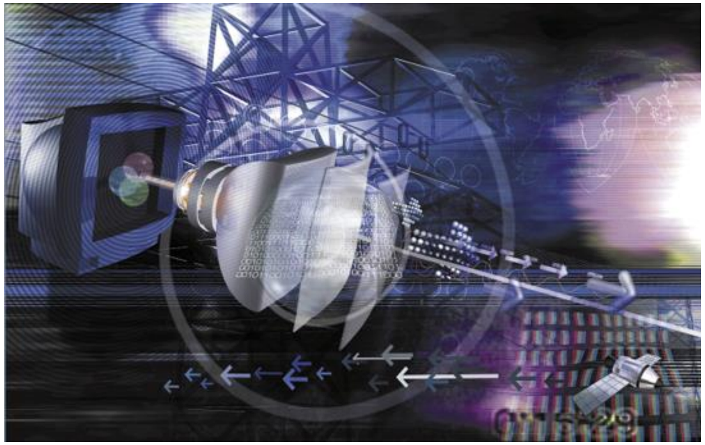
Communication Circuits Design

Academic year 2018/2019 – Semester 2 – Week 4

Lecture 4.3: System design examples

Francesco Fioranelli, Wasim Ahmad, Faisal Tariq

Outline



- **A comprehensive series of exercises essentially**

References:

- *R. Sobot, "Wireless Communication Electronics", Springer, at UoG Library online – Chapter 13 (mostly taken from this)*
  - *Note the companion book "Wireless Communication Electronics by Example", which is also available at UoG online library with solutions to all exercises*
- *J. Beasley, G. Miller, "Modern Electronic Communication", Pearson, 9th ed. – Pages 150, and 300 and following*

Notes

These slides contain a lot of exercises. Some of them are rather straightforward and short. Some of them are long and more complex. Please use these slides as an opportunity to review all the relevant material.

We aim to do in class together examples 1 2 4 7 9 11

The others are left for your own individual study and some will be discussed at the tutorial session.

Do look at the example of “mock” exam paper and solve the exercises in there.

What do I need to know?

As a good engineer you should be able to…

- **Explain** (in English)…explain and describe what something is or what something does, what its properties are

For example, explain the role of a mixer in an RF heterodyne receiver and list some of the key properties and metrics of a mixer

- **Characterise**…use your knowledge and some maths to show that you understand a certain component or system

For example, given two inputs signals to the mixers and a complementary image rejection filter, calculate the spectrum of the output and the level of the image frequency

- **Design**… given some requirements, either modify an existing circuit/component/system to meet them, or select/combine different choice to meet such requirements

For example, given a certain requirements in terms of RF/IF/LO frequencies and power level, select among possible mixers options and discuss pros and cons of those

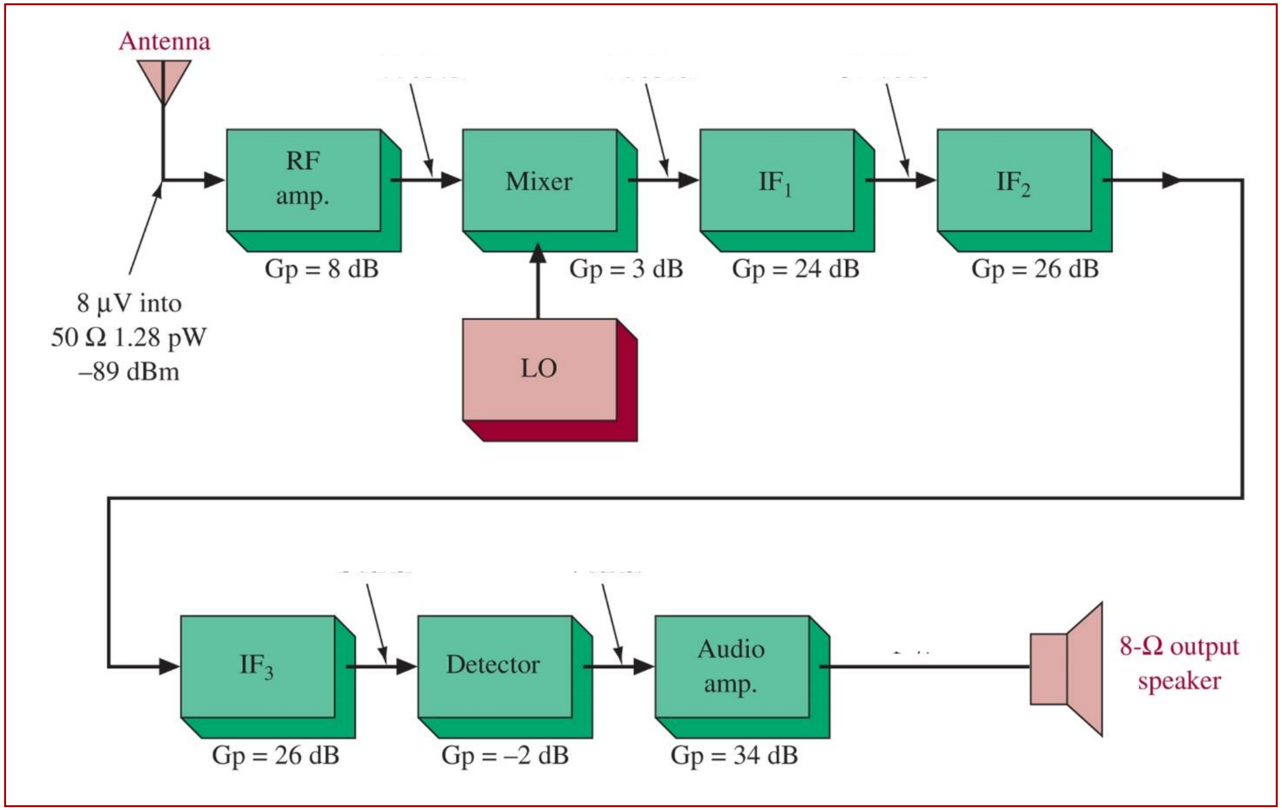
This is more or less the structure of an exam question…

EX1

Consider the radio receiver shown below. The antenna receives a 8μV signal at its 50Ω input impedance.

Calculate the input power to the receiver in Watt and in dBm. Calculate the power driven into the speaker. Note the processing gain Gp of each block (IF are amplifiers at IF frequency).

Hint: calculate the power given voltage and impedance, convert in dBm, add/subtract the processing blocks' contribution in dB



EX1 - Solution

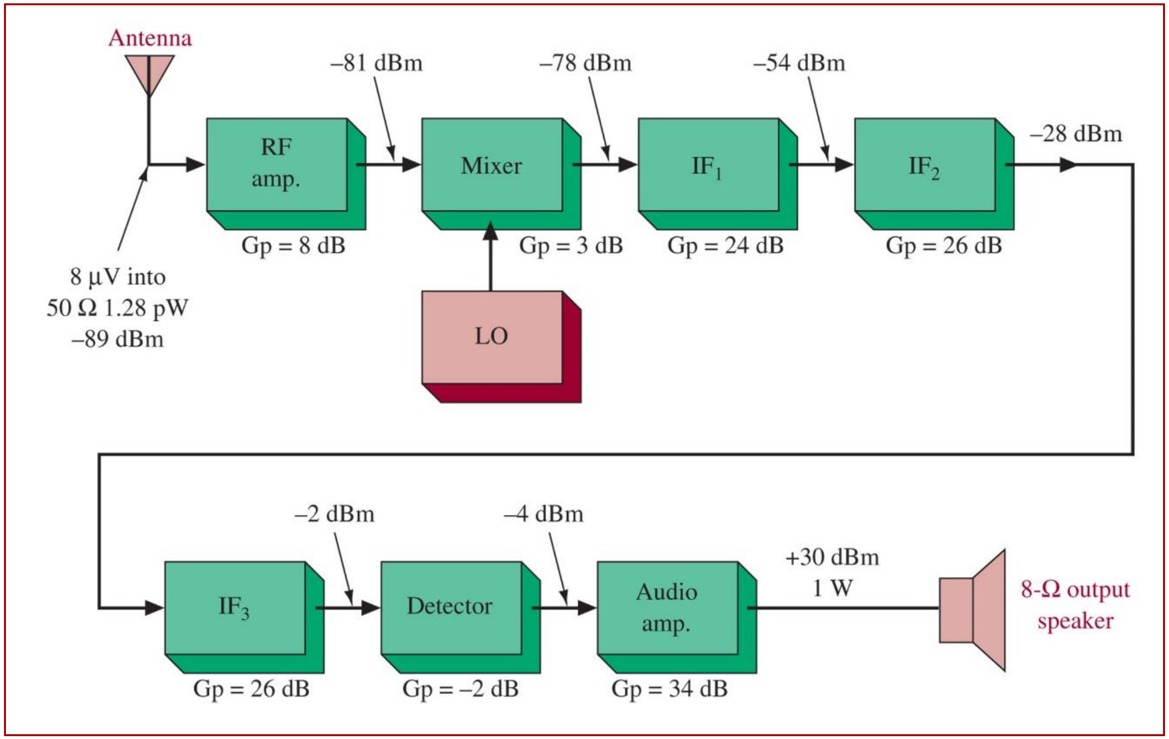
First calculate the input power going from the antenna into the receiver.

$$P = \frac{V^2}{R} = \frac{(8\ \mu\text{V})^2}{50\ \Omega} = 1.28 \times 10^{-12}\ \text{W}$$

$$\begin{aligned}\text{dBm} &= 10\log_{10}\frac{P}{1\ \text{mW}} \\ &= 10\log_{10}\frac{1.28 \times 10^{-12}}{1 \times 10^{-3}} = -89\ \text{dBm}\end{aligned}$$

Then add gain/losses of each block down to the speaker

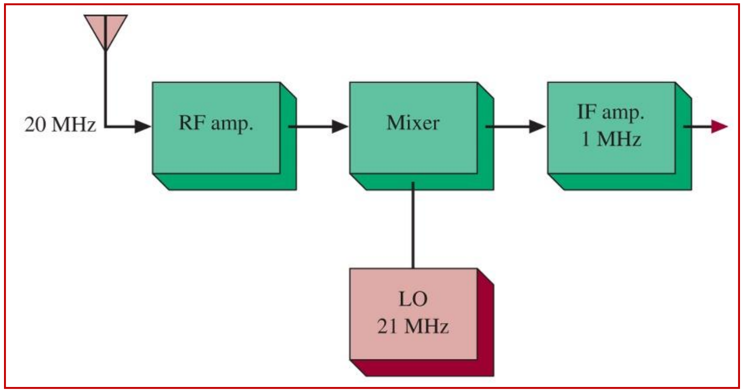
$$\begin{aligned}P_{\text{out(dBm)}} &= -89\ \text{dBm} + 8\ \text{dB} + 3\ \text{dB} + 24\ \text{dB} + 26\ \text{dB} \\ &\quad + 26\ \text{dB} - 2\ \text{dB} + 34\ \text{dB} \\ &= 30\ \text{dBm into speaker}\end{aligned}$$

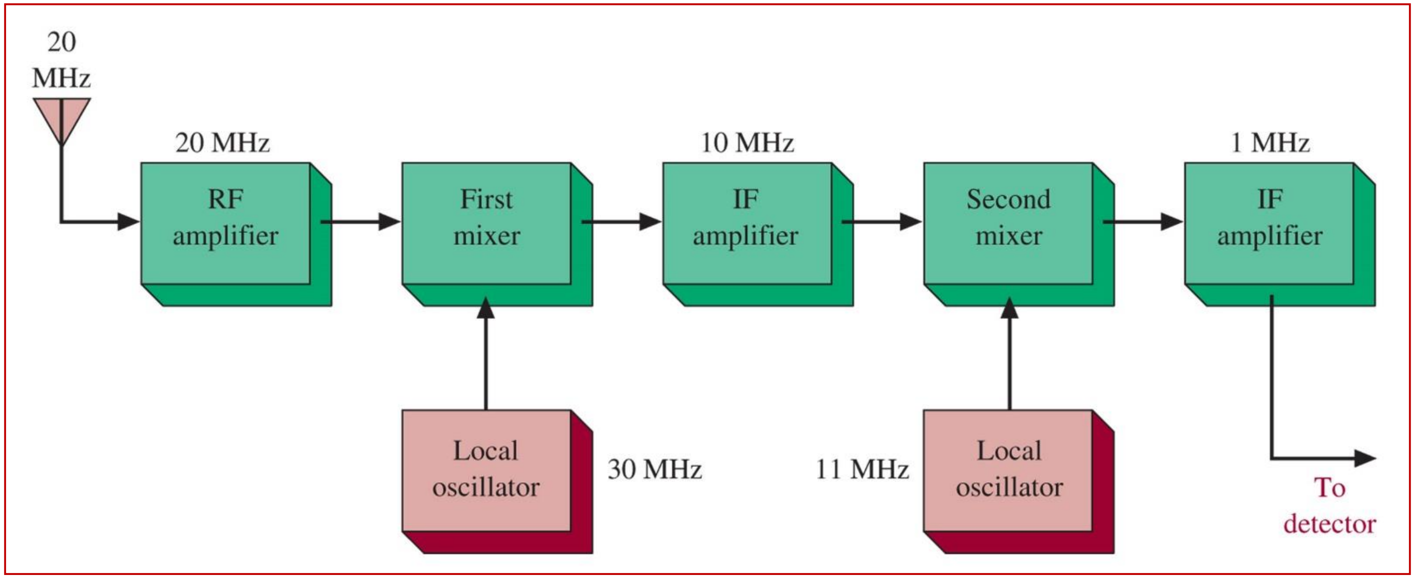


EX2

Determine the image frequency for both receivers shown below.

Remember, the image frequency is such to generate an output of the first mixer equal to the desired IF signal (1 MHz in both cases).

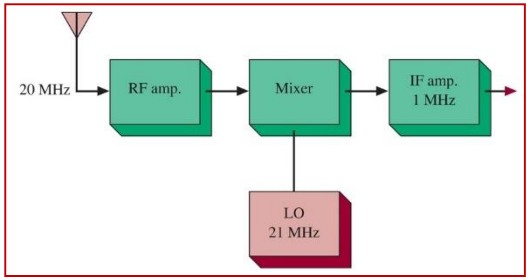


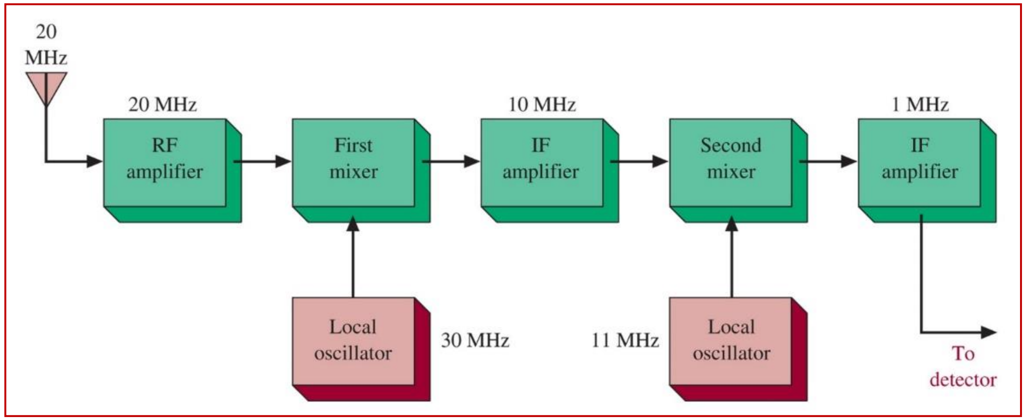


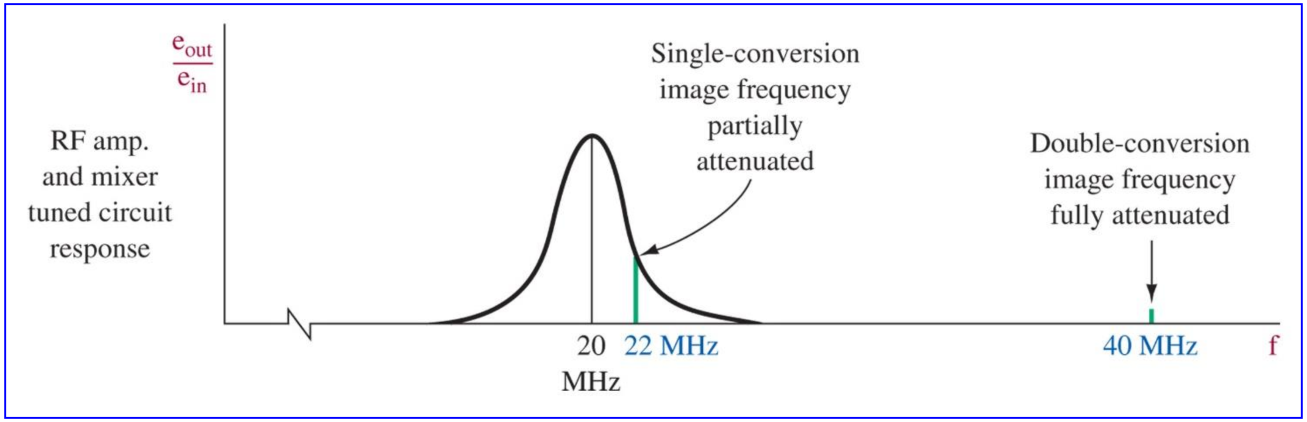
EX2-Solution

It should be straightforward to see that in the first case the image frequency is 22 MHz (rather close to the desired received frequency of 20 MHz, hence hard to filter out), whereas in the second case the image is at 40 MHz (farther away from the desired frequency of 20 MHz).

This should show to you why it is advantageous to use double-conversion in superheterodyne receivers for better image rejection. Check also the spectra.

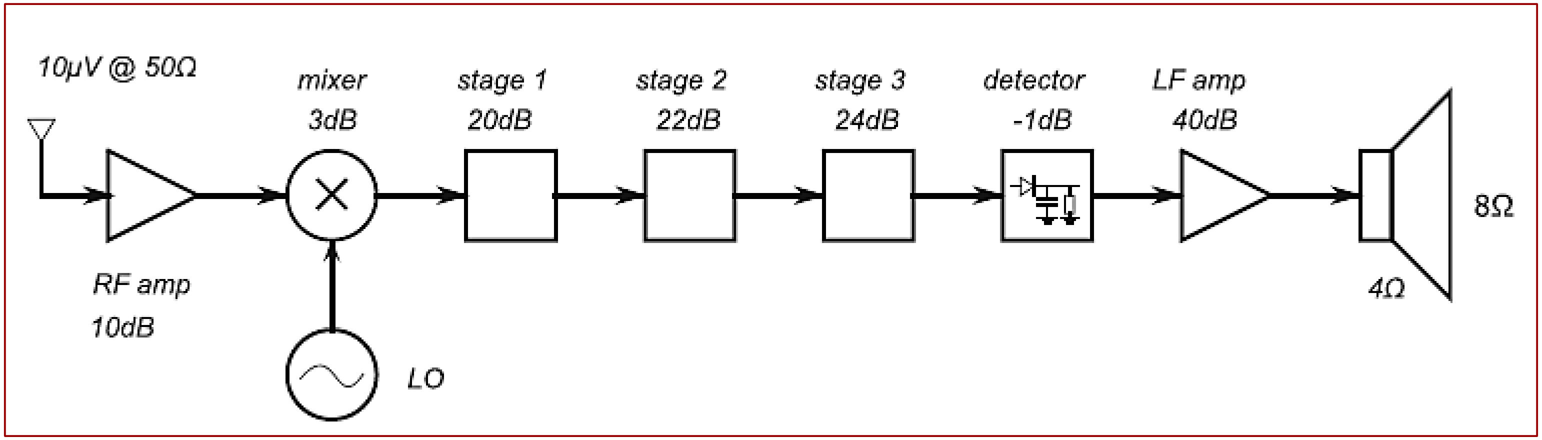






EX3 with solution

The voltage signal received by a 50Ω antenna has an amplitude of 10μV. Looking at the receiver system shown below, calculate the input signal power in dBm and Watt and the power delivered to speaker in dBm and Watt.



This is very similar to EX1 so just the numerical solution is provided here.
Pin = 2pW corresponding to roughly -87dBm
Pspeaker = +21dbm
If you remember (or calculate) that 30dBm = 1 Watt, then + 21dBm= -9dBWatt
Solving $-9dBWatt = 10\log_{10} Pspeaker\ in\ Watt$

$$\rightarrow Pspeaker\ in\ Watt = 10^{-9/10} \cong 0.13\ Watt$$

EX4

An AM receiver is designed to receive RF signals in the 500-1600 kHz frequency range with a required bandwidth of BW = 10 kHz at f_0=1050 kHz. The RF amplifier uses a fixed inductor L=1μH.

Q1- Calculate the bandwidth and the required capacitance C at fmax=1600 kHz and at fmin=500kHz. Assume that the quality factor Q is constant across the range of frequencies of interest.

Tips: First calculate the quality factor at the central frequency f_0, then calculate the bandwidth at the max/min frequencies and the related capacitance (assuming the circuit is resonating also at max/min frequency).

Q2 – What is the minimum channel spacing Δf between adjacent AM channels to prevent inter-channel interference with this AM receiver?

Tip: think that, in order to avoid interference, the bandwidth of the tuned receiver must be always smaller than the spacing between channels

EX4 - Solution

Q factor

By definition, Q factor of an LC resonator is calculated as

$$Q = \frac{f_0}{B} = \frac{1050\,\text{kHz}}{10\,\text{kHz}} = 105$$

At the max frequency

Assuming constant Q factor at $f_0 = 1600\,\text{kHz}$ the achieved bandwidth is

$$B_{fmax} = \frac{f_{max}}{Q} = \frac{1600\,\text{kHz}}{105} = 15.238\,\text{kHz}$$

which requires capacitance

$$C = \frac{1}{(2\pi\ f_{max})^2\,L} = 9.895\,\text{nF}$$

…and at the min frequency

Similarly, at $f_{min} = 500\,\text{kHz}$ for the given data we fand

$$B_{fmin} = \frac{f_{min}}{Q} = \frac{500\,\text{kHz}}{105} = 4.762\,\text{kHz}$$

which requires that the tuneable capacitor is set to

$$C = \frac{1}{(2\pi\ f_{min})^2\,L} = 101.321\,\text{nF}$$

Q2 -> The largest bandwidth that this AM receiver can process is 15.238 kHz when the carrier frequency is high (1600 kHz). So the minimum spacing between channels must be set to be $\Delta f \geq BWmax = 15.238kHz$

EX5 with solution

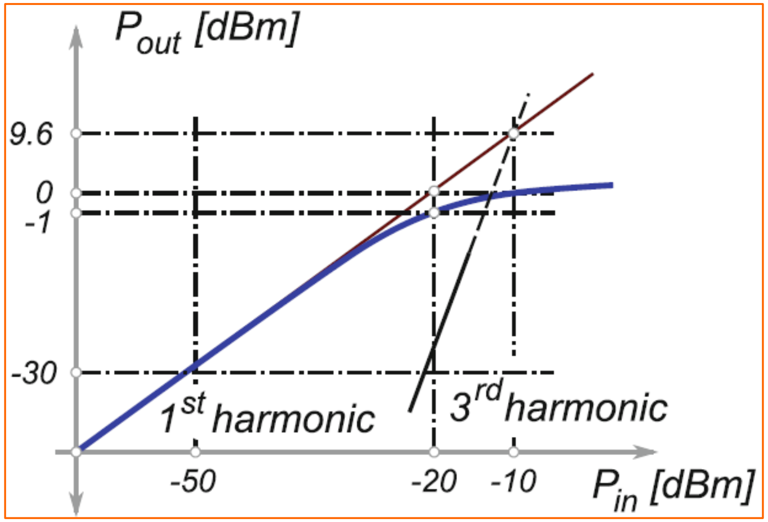
Q1- The LO oscillator frequency in a receiver is 11 MHz and the RF signal frequency is 10 MHz. What is the image frequency and what is the desired frequency resulting from the mixing process?

Tips- Recall that at the receiver you aim to down-convert the RF signal to a lower frequency IF signal, and recall that the mixer will produce the sum & difference components…
Almost too easy to write the solution!

Q2- Imagine this mixer is a an ideally driven level 10 mixer. If the isolation LO to IF is 45 dB, calculate the power level in dBm of the LO leakage at the IF port of the mixer.
Tips- Recall what the level of a mixer means and this is an ideally driven mixer. Recall the meaning of the isolation parameter…

EX6 with solutions

The Input/Output power characteristic of an amplifier is shown below. Explain the meaning and estimate the values of the parameters Gain, 1dB Compression Point, and IIP3.



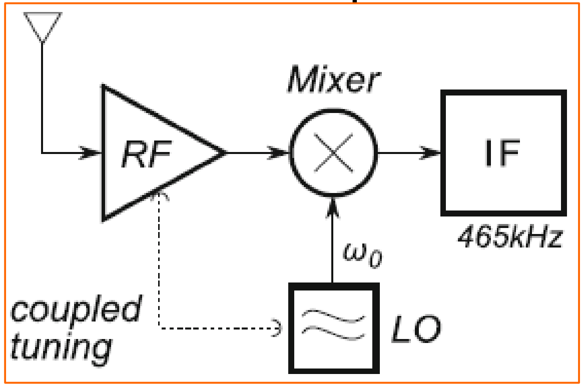
Gain – defined in the linear region as the ratio output over input (remember in dB a ratio is a difference for properties of logarithms) {20 dB}

1dB Compression Point – The input level at which the actual output is 1dB below the ideal linear trend {-20 dBm}

IIP3 – The input level at which the ideal power of the first harmonic and third order harmonic intercepts – in this case {-10dBm} for the input IIP3 and {+9.6dBm} for output OIP3

EX7

An AM receiver is designed to receive RF signals in the 500-1600 kHz frequency range. All RF incoming signals are shifted to IF = 465 kHz. The receiver is tuned using a knob that simultaneously changes the capacitance of the RF amplifier and Local Oscillator as shown in the figure.



Calculate:

1. The tuning ratio $C_{RF}(max)/C_{RF}(min)$ of the capacitor in the RF amplifier's resonator

2. The tuning ratio $C_{LO}(max)/C_{LO}(min)$ of the capacitor in the local oscillator's resonator

3. As the LO could be tuned to be higher or lower in frequency than the RF signal, recommend what frequency is <u>more suitable from a practical engineering perspective</u>.

Tips – Note that both the RF amplifier and the LO have LC tuning/matching networks that act as band-pass filters to tune in at the desired frequency. The tuning ratio looks at the ratio of the values of C for the max and min frequency. Consider a fixed inductor L

EX7 with solution

1. First RF part, where the max and min frequencies are given as 500-1600 kHz

$$R_{fRF} = \frac{f_{RF}(max)}{f_{RF}(min)} = \frac{1600\,\text{kHz}}{500\,\text{kHz}} = 3.2$$

$$\therefore$$

$$\frac{f_{RF}(max)}{f_{RF}(min)} = \frac{\frac{1}{2\pi\sqrt{LC_{min}}}}{\frac{1}{2\pi\sqrt{LC_{max}}}} = \sqrt{\frac{C_{max}}{C_{min}}}$$

$$\therefore$$

$$R_{cRF} = \frac{C_{RF}(max)}{C_{RF}(min)} = \left[\frac{f_{RF}(max)}{f_{RF}(min)}\right]^2 = 10.24$$

2. Then the LO part. For this in general this relation is valid $f_{IF} = |f_{RF} - f_{LO}|$ which can have 2 cases. Recall the RF frequency has min/max values from above

   a) $f_{LO} > f_{RF}$

$$f_{LO}(min) = f_{RF}(min) + f_{IF} = 500\,\text{kHz} + 465\,\text{kHz} = 965\,\text{kHz}$$

$$f_{LO}(max) = f_{RF}(max) + f_{IF} = 1600\,\text{kHz} + 465\,\text{kHz} = 2065\,\text{kHz}$$

Hence, the local oscillator's frequency tuneability ratio R_{fLO} and capacitor tuneabilility ratio R_{cLO} are

$$R_{fLO} = \frac{f_{LO}(max)}{f_{LO}(min)} = \frac{2065\,\text{kHz}}{965\,\text{kHz}} = 2.1399$$

$$\therefore$$

$$R_{cLO} = \frac{C_{LO}(max)}{C_{LO}(min)} = \left[\frac{f_{LO}(max)}{f_{LO}(min)}\right]^2 = 4.5792 \qquad (24.3)$$

EX7 with solution

2. LO part continuation.

   b) $f_{LO} < f_{RF}$

$$f_{LO}(min) = f_{RF}(min) - f_{IF} = 500\,\text{kHz} - 465\,\text{kHz} = 35\,\text{kHz}$$
$$f_{LO}(max) = f_{RF}(max) - f_{IF} = 1600\,\text{kHz} - 465\,\text{kHz} = 1135\,\text{kHz}$$

Hence, the local oscillator's frequency tuneability ratio R_{fLO} and capacitor tuneabilility ratio R_{cLO} are

$$R_{fLO} = \frac{f_{LO}(max)}{f_{LO}(min)} = \frac{1135\,\text{kHz}}{35\,\text{kHz}} = 32.429$$
$$\therefore$$
$$R_{cLO} = \frac{C_{LO}(max)}{C_{LO}(min)} = \left[\frac{f_{LO}(max)}{f_{LO}(min)}\right]^2 = 1051.6 \qquad (24.4)$$

3. Which LO approach is most suitable among the case a and case b? From a practical perspective you need to consider that manufacturing capacitors and LOs with wide tuning ratios is more difficult than those with smaller ranges.

In case a), the capacitor tuning ratio is around 4.6 and the tuning range of the LO about 2.1, whereas in the second case the values are much larger, about 1050 for the capacitor and 32.4 for the LO.

The first case, case a) is therefore a much more practical engineering solution.

EX8

A medium wave AM transmitter operates in the 540-1610 kHz range with 10 kHz spacing while using 455 kHz IF frequency.
Estimate the range of the LO frequencies and suggest band-pass filter(s) for use with a possible AM medium wave receiver.

Tip- Consider what information you need and the usual block diagram of receiver. You basically have an RF signal of bandwidth 10 kHz that can be between 540-1610 kHz and you want to down-convert that to a fixed frequency of 455 kHz

Regarding the suggestion of possible BPF, you need to estimate the necessary Q given the bandwidth and the carrier frequency, and that will have implications on the L & C required. However, where would you put the BPF?
-One BPF for each 10 kHz channel at the RF part?
-One BPF for 10 kHz at the IF part?
-Or both?

EX8 with solution

Classic mixer theory will provide the LO frequency which can generate the required 455 kHz signal, at both the upper and lower RF frequency.

$$f_{LOmin} = f_{IF} + f_{RFmin} = 455\,\text{kHz} + 540\,\text{kHz} = 995\,\text{kHz}$$
$$f_{LOmax} = f_{IF} + f_{RFmax} = 455\,\text{kHz} + 1610\,\text{kHz} = 1965\,\text{kHz}$$

To find the Q, you can apply its definition given the bandwidth 10 kHz and upper/lower RF frequencies.

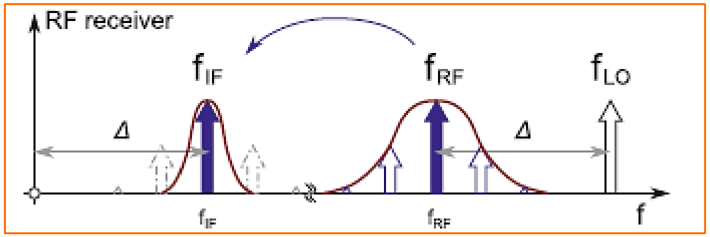
If you look for filters across the whole RF frequency range, then max/min Q is

$$Q_{max} = \frac{f_{RFmax}}{B} = \frac{1610\,\text{kHz}}{10\,\text{kHz}} = 161$$
$$Q_{min} = \frac{f_{RFmin}}{B} = \frac{540\,\text{kHz}}{10\,\text{kHz}} = 54$$

However, a filter with a tunable Q of that wide range is hard to achieve. The best approach is to have a low-Q filter at the RF to let all relevant RF frequencies into the mixing stage (Q=54), then design a more selective filter at the fixed IF 455 kHz

$$Q = \frac{f_{IF}}{B} = \frac{455\,\text{kHz}}{10\,\text{kHz}} = 45.5$$

The figure below shows the spectral representation of the two filters – the filter at IF has fixed Q so it is much simpler to design than a tunable filter!



EX9

A double conversion reveiver architecture is based on two IF frequencies, IF1 at 10.7 MHz and IF2 at 455 kHz. If the receiver is tuned to a 20 MHz RF signal, find the frequencies of the LO and the values of the undesired sum/difference harmonics from the mixing stages.

Tip- Double conversion means superheterodyne receiver, where the RF signal is first down-converted to the first IF1, then again converted to the second, lower IF2. Regarding the undesired harmonics, remember that the mixer output has (as a simple approximation) sum/difference harmonics of the input signals.

EX9 with solution

For the first down-conversion stage, you want to go from 20 MHz RF to 10.7 MHz IF – It is fairly straightforward to see that the required LO1 frequency is 30.7 MHz

$$f_{IF1} = f_{RF} \pm f_{LO1} \;\rightarrow f_{LO1} = f_{RF} + f_{IF1} = \mathbf{30.7\ \mathit{MHz}}$$

Note that you could have also chosen 9.3 MHz, but **it is always practical to set the LO frequency higher than the RF input frequency if possible, for better separation of the various spectral components.**

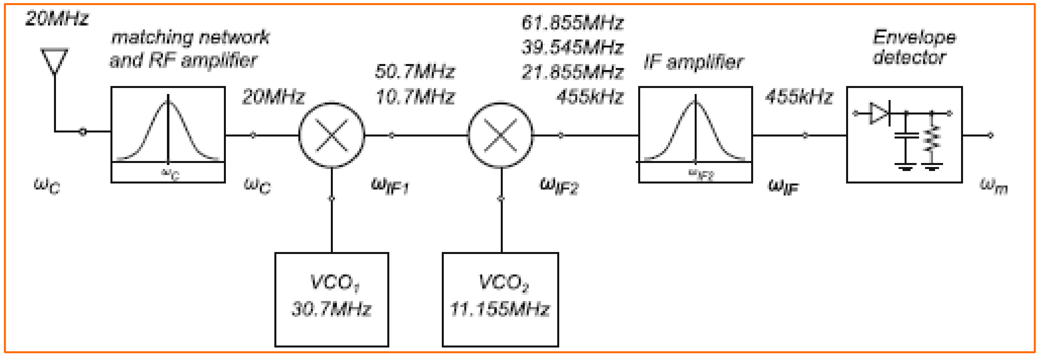
Choosing 30.7 MHz as LO1 will also generate at the output the undesired component 50.7 MHz.

For the second down-conversion stage, focus on 10.7 MHz input and desired IF2 output 0.555 MHz. Repeating the same process as before to find LO2, you should use $f_{LO2} = f_{IF1} + f_{IF2} = \mathbf{11.155\ \mathit{MHz}}$ (again go for LO higher than the RF input)

This will also generate the unwanted components 21.855 MHz.

However, at the input of the second stage, you also have the undesired output of the first stage 50.7 MHz – combining this with the LO2 found above, you have two additional output frequencies 61.855 MHz and 39.545 MHz

Make sure you fully understand this diagram!!!



EX10

12.6. A receiver whose IF frequency is $f_{IF} = 455\,\text{kHz}$ is tuned to an RF signal with $f_{RF} = 950\,\text{kHz}$. No other transmission frequency is allowed within the RF band $f_{RF} = 950\,\text{kHz} \pm 10\,\text{kHz}$. However, for the sake of argument, let us imagine existence of a nearby non–linear transmitter whose emitting frequency spectrum consists of its both first and the second harmonics.

Aside from the obvious $950\,\text{kHz} \pm 10\,\text{kHz}$ frequency range, what other frequency range(s) should also be prohibited for the external transmitter?

Taken from *"Wireless Communication Electronics by Example",* which is also available at UoG online library with solutions to all exercises.

Tip – When we talk about forbidden/prohibited frequencies in a receiver, we are referring to problems of <u>image frequency rejection</u>. In this case you have an RF signal at 950±10 kHz which is down-converted to 455±10 kHz.

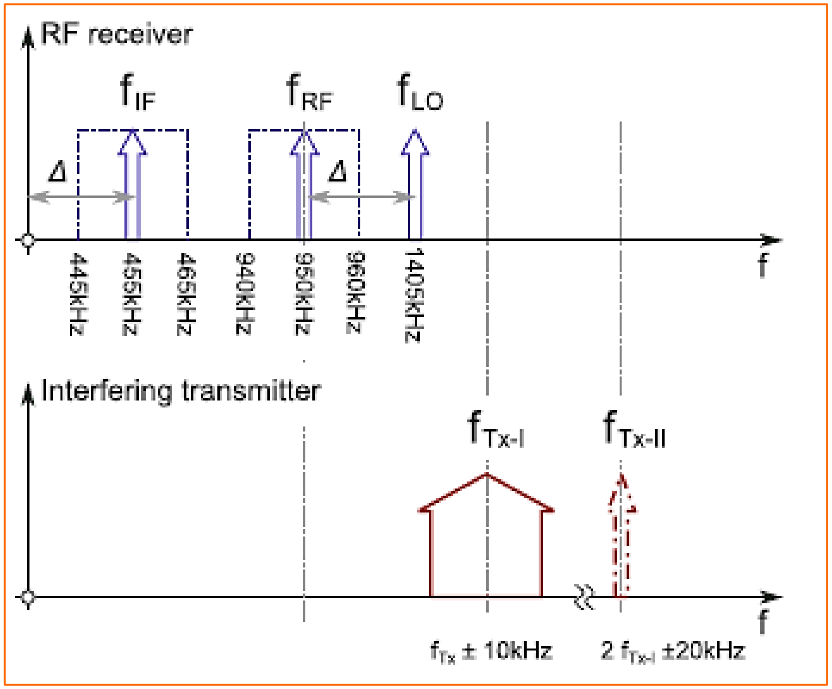
Nobody is interfering at that RF frequency, but what if the harmonics of a "dirty" transmitter are, in particular the 2nd harmonic?

EX10 with solutions

So interfering transmitters cannot work in the 950±10 kHz band.

-They could work at higher frequencies. In this case both the first harmonic f_{TX-I} and the second harmonic f_{TX-II} (twice the first), would be far away from the RF band of the receiver.

In this case these interfering signals would be filtered out by the LC bandpass filter at the RF input of the mixer stage (tuning network)



EX10 with solutions

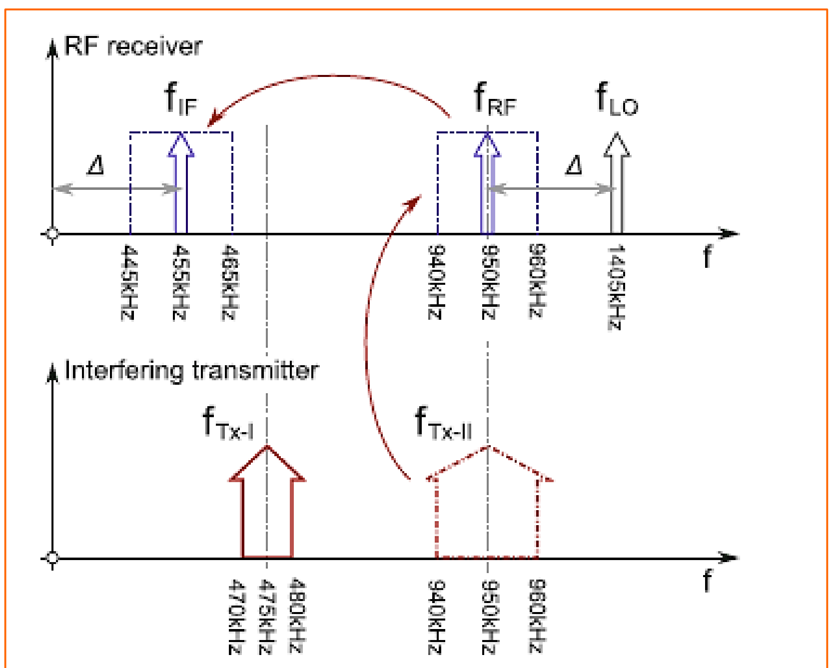
So interfering transmitters cannot work in the 950$\pm$10 kHz band.

-However they could be operating below the desired RF band. In this scenario, the worst case would be when the second harmonic $f_{TX\text{-}II}$ is overlapping with the desired RF band.

Bear in mind that the second harmonic is twice the fundamental, first harmonic, so mathematically

$$f_{TX-II} = 2f_{TX-I} = 950 \pm 10kHz$$

Hence $\boldsymbol{f_{TX-I} = 475 \pm 5kHz}$ This would be the additional forbidden range of frequencies for transmission.

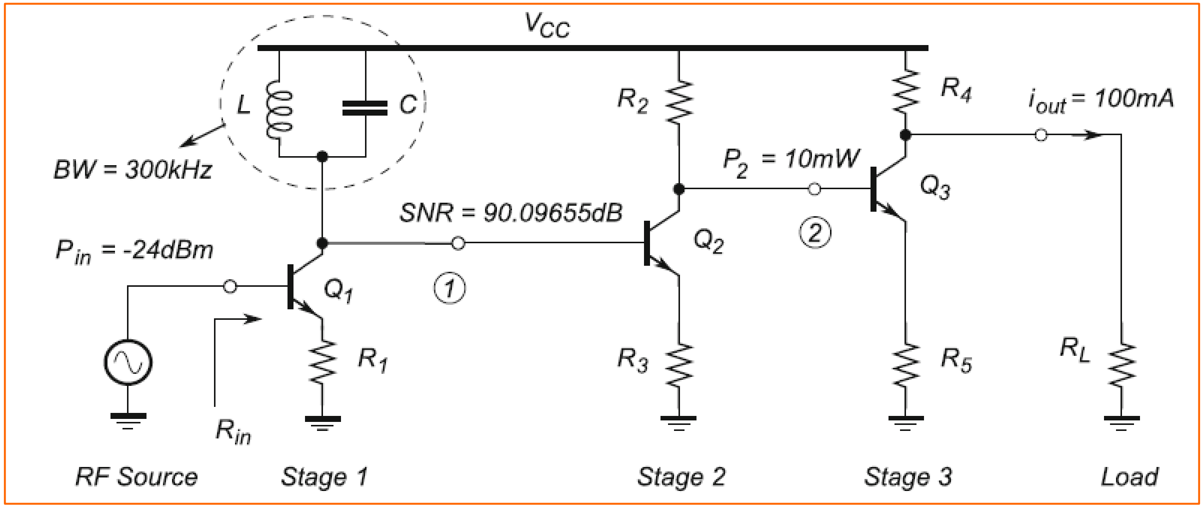


EX11

12.11. Conceptual and simplified schematic diagram of a three stage RF amplifier is in Fig. 12.6. Power level P_{in} of the RF source signal that corresponds to the 1dB compression point, signal–to–noise ratio SNR measured at the internal node ①, signal power level P_2 measured at the internal node ②, and the current i_{out} of the signal delivered to the load R_L are shown explicitly. In addition, bandwidth of the LC resonator is B is set by the design of RF amplifier's LC resonator.

1. *Data*: $R_L = 100\,\Omega$, temperature is $T = 26.85\,°\mathrm{C}$, power gain of the first stage is $A_1 = 10\,\mathrm{dB}$, noise figure of the second stage is $NF_2 = 6.02\,\mathrm{dB}$, and noise figure of the third stage is $NF_3 = 9.0309\,\mathrm{dB}$,
2. *Assumptions*: the total input side noise is generated only by the equivalent input resistance R_{in} while each gain stage generates its own noise, the load resistor is ideal, all base currents are ignored. Using the given data and assumptions, estimate:

   a. power gain of the second stage only A_2 in [dB];
   b. noise voltage v_n at the output;
   c. noise figure NF of this amplifier;
   d. dynamic range DR of this amplifier.

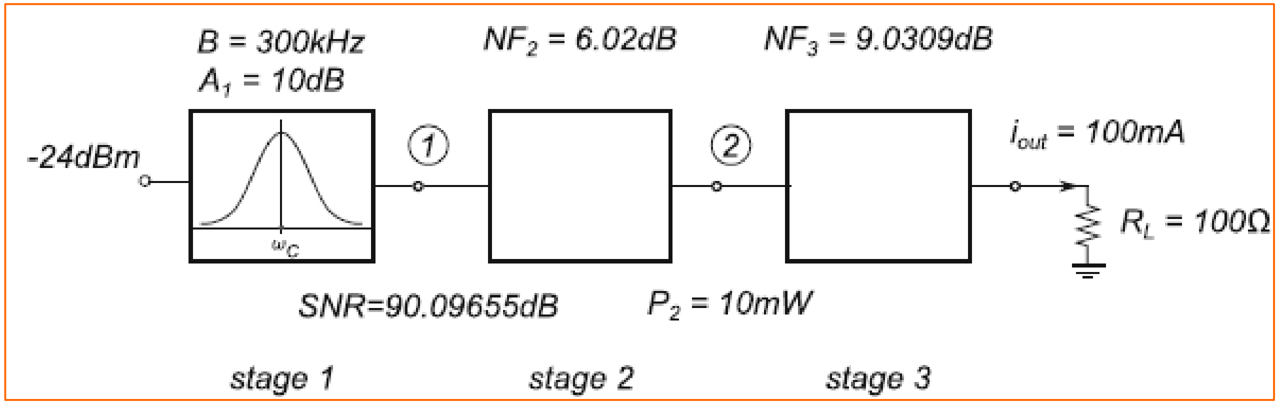


EX11 with solutions

Tip- Before diving in into the calculations you need to be able to think in terms of block diagrams, i.e. system approach (each block provides losses or gain to input/output powers) vs circuit approach (solve the circuit applying Kirchhoff & co).

If you look at the circuit you can see three transistors that act as 3 amplifier blocks, plus a source and a load. The initial stage has also a tuning LC circuit

So what we get is this



A. **Power gain of 2nd stage in dB?** This is equal to the power at 2nd stage P2 minus the power at the 1st stage P1.

We know P2=10mW=10dBm

We can find P1=input + gain of 1st stage A1 = -24dBm+10dB= -14dBm
So A2=P2-P1=10dBm-(-14dBm)=**24dB**

EX11 with solutions

B. Noise voltage v_n at output? The power on the output resistance can be written as $P = V^2/R_L$, so you can use this formula but need to find the noise power at the output $P_{n,out}$.

If you look at the circuit, this is a 3-stages amplifier so the output noise power will be the amplified input noise power $A_{tot}P_{n,in}$ plus the total noise figure NF_{tot} of the system. So we need to find these 3 quantities.

For the input noise power, this is a simple formula but recall that the you have an RLC band-pass filter at the beginning, hence use the effective bandwidth

$$P_{n,in} = kT\Delta f_{eff} = k\,(26.85 + 273)\,\frac{\pi}{2}\,300000 = 1.95 \times 10^{-15}W$$

Converting into dBm $10\log_{10}\left(\frac{1.95\times10^{-15}}{0.001}\right) = -117.1\,dBm$

For the total gain, you found the gain of the 1st stage in the previous question and the gain of the 2nd stage is given. What about 3rd stage?

Well, we know its input power P_2=10mW=10dBm. We can find the output power knowing R_L and the current as $P_{out} = I_{out}^2 R_L = 1W = 30dBm$. So the gain of the 3rd stage is 20dB.

The total gain of the 3-stages system will therefore be

$$A_{tot} = A_1 + A_2 + A_3 = 10 + 24 + 20 = 54\,dB$$

EX11 with solutions

Now we need to find the total noise figure NF_{tot}. This is the total noise factor F_{tot} measured in dB. One can find the total noise factor F_{tot} by using Friis formula for cascaded system $F_{tot} = F_1 + \frac{F_2 - 1}{A_1} + \frac{F_3 - 1}{A_1 A_2}$

We know the F or NF of the 2nd and 3rd stage, only the 1st one is missing. We recall that the noise figure is the difference in SNR between input and output so that $NF_1 = SNR_{in} - SNR_{out}$

We know the SNR at the output. To find the SNR at the input we know the signal input power (given by the problem) and we found the noise input power (see in previous slide). Hence $SNR_{in} = P_{s,in} - P_{n,in} = +93.1\, dB$ and $NF_1 = 93.1 - 90.096 = 3dB$

Now, we have NF for all stages, we can convert into noise factors F and use Friis formula

$NF_1 = 3.010\,\text{dB}$	$\therefore$	$F_1 = 2$
$NF_2 = 6.020\,\text{dB}$	$\therefore$	$F_1 = 4$
$NF_3 = 9.030\,\text{dB}$	$\therefore$	$F_1 = 8$

so $F_{tot} = F_1 + \frac{F_2 - 1}{A_1} + \frac{F_3 - 1}{A_1 A_2} = 2 + \frac{3}{10} + \frac{7}{2511} = 2.303$

and $NF_{tot} = 10\log_{10}2.303 = 3.622dB$

Finally we can find $P_{n,out} = P_{n,in} + A_{tot} + NF_{tot} = -117.1 + 54 + 3.622 = -59.48dBm$

We can convert this into voltage $P_{n,out} = 10^{-59.48/10} = 1.13 \times 10^{-6}mW = 1.13 \times 10^{-9}W$

And finally convert from power to voltage so $V_{n,out} = \sqrt{P_{n,out} R_L} = 3.36 \times 10^{-4}V = \mathbf{336\mu V}$

C. NF of the amplifier. We have implicitly answered this in the previous point, i.e. **3.622dB** for the total noise figure.

EX11 with solutions

D. The dynamic range of the amplifier system.

We saw that the dynamic range DR is given by the input power level (typically at 1dB compression point) minus the sensitivity. This in turn is the total noise floor of the system (accounting for the effective bandwidth and total NF) plus the SNR. So:

$$P_n = -174dBm + 10log_{10}(\Delta f_{eff}) + NF = -174 + 56.7 + 3.66 = -113.64dBm$$

$$S_n = P_n + SNR_{out} = -113.64 + SNR_{out}$$

But we need the SNR_{out}. We know the output signal power and the output noise power so we can calculate $SNR_{out} = P_{s,out}$ - $P_{n,out} = 30dBm - (-59.48) = 89.48dBm$. Hence $S_n = P_n + SNR_{out} = -113.64 + 89.48 = -24.16dBm$

So finally the dynamic range can be calculated

$$DR = 1dB_{point} - S_n = -24dBm - (-24.16dBm) \cong \mathbf{0}\boldsymbol{dB}$$

!!This essentially means that this amplifier is not performing well, because its dynamic range is close to zero, hence no useful signals can be processed with it!!

EX12

A mobile phone receiver operates at temperature 290K and has noise figure NF=20dB, bandwidth 1 MHz, and SNR=0dB. Also, note that the receiver exhibits a non-linearity that can be described as $y(x) = 2x - 0.267x^3$ where y(x) is the output signal and the incoming signal can be described as $x_1 = 1sin(\omega_1 t)\, V$

Calculate:

1. The 1dB compression point in dB
2. The 3rd order input intercept point IIP3 in dB
3. The sensitivity in dBm
4. The dynamic range in dB

Tips – this is a problem involving non-linearity, where the input signal is a sinusoidal signal in this case, and the output has third harmonics. Rememeber we saw formulae to relate the parameters 1dB compression point and IIP3 to the coefficients of the non-linear relation.

EX12 with solutions

1. The 1dB compression point can be found with this relationship below

$$P_{1dB} = \sqrt{0.145\frac{|a_1|}{|a_3|}} = \sqrt{0.145\frac{|2|}{|0.267|}} = 1.04 \cong \mathbf{0.18}\boldsymbol{dB}$$

2. The IIP3 point can be found with this relationship below

$$IIP3 = \sqrt{4/3\frac{|a_1|}{|a_3|}} = \sqrt{4/3\frac{|2|}{|0.267|}} = 3.16 \cong \mathbf{5}\boldsymbol{dB}$$

3. The sensitivity is given by the true noise floor (accounting for the effective bandwidth and total NF) plus the SNR

$$S_n = -174dBm + 10log_{10}\left(\Delta f_{eff}\right) + NF + SNR_{out} = -174 + 10log_{10}(10^6) + 20 + 0$$
$$= \boldsymbol{-94dBm}$$

4. The dynamic range can be defined comparing the sensitivity with the input power, either at 1dB compression point or at the IIP3 point. Hence

$DR = 1dB_{point} - S_n = 0.18 - (-94) = \boldsymbol{93.82dB}$ OR

$DR = IIP3 - S_n = 5 - (-94) = \boldsymbol{99dB}$