# COURSE OUTLINE

- Introduction
- Signals and Noise
- Filtering: Band-Pass Filters 1 - BPF1
- Sensors and associated electronics

# Band-Pass Filters 1



- Narrow-Band Signals
- Recovering Narrow-Band Signals from Noise
- Moving Signals in Frequency (Signal Modulation)

# Narrow-Band Signals

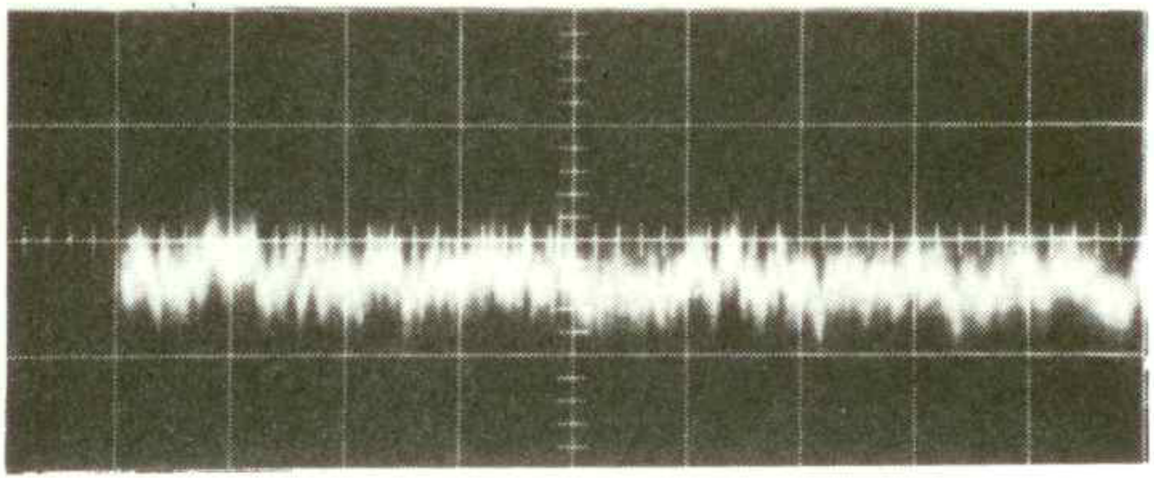
# Narrow-Band Signals

**Power signals** with a narrow power spectrum, that is, a peak with

- center-frequency $f_s$
- bandwidth $\Delta f_s$ which is small in absolute value, typically $\Delta f_s < 10$ Hz, and/or with respect to the center frequency $\Delta f_s \ll f_s$

They approximate well a sinusoid over a wide time interval $T_s \approx 1/\Delta f_s$

**QUESTION**: how can we measure such narrow-band signals in presence of intense white noise? And what if also 1/f noise is present?



# Recovering Narrow-Band Signals from Noise

# Recovering Narrow-Band Signals from noise



Let's see some typical examples of signals with

- narrow linewidth $\Delta f_s = 1\ Hz$
- small amplitude $V_s \leq 100\ nV$

for bringing them to higher level (suitable for processing circuits: filters, meters, etc.) they are amplified by a DC-coupled wide-band preamplifier with

- upper band-limit $f_h = 1MHz$
- noise spectral density (referred to input) with
  «white» component $\sqrt{S_b} = 5nV/\sqrt{Hz}$
  and *1/f* component with corner frequency $f_c = 2kHz$

Let us consider three cases with different center-frequency $f_s$ :

- Case 1: **high** frequency $f_s$ = 100 kHz
- Case 2: **moderately low** frequency $f_s$ = 1 kHz
- Case 3: **low** frequency $f_s$ = 10 Hz

# Recovering Narrow-Band Signals



**CASE 1: signal $V_s \leq 100\ nV$ at high frequency $f_s$ = 100 kHz**

**a) observing the voltage waveform in the TIME DOMAIN, i.e. on oscilloscope display**

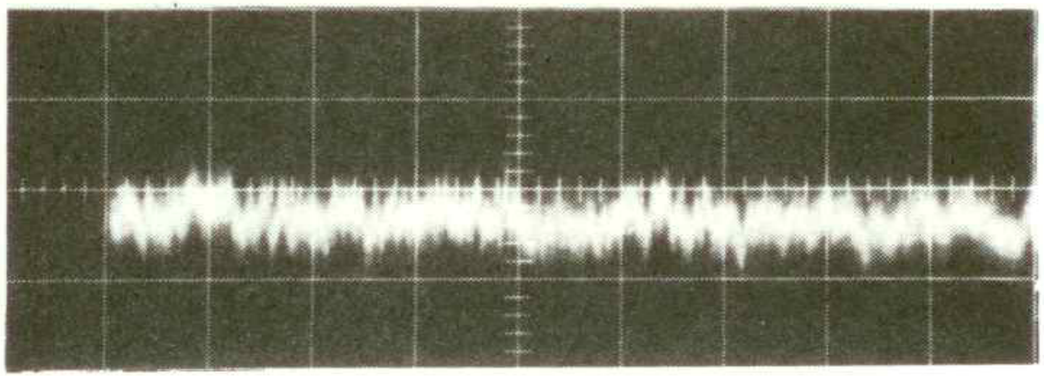
The signal to be recovered is at frequency $f_s$ = 100 kHz much higher than the noise corner frequency $f_c = 2kHz$ , so that we can use a simple high-pass filter with band-limit $f_i$ = 10kHz to cut off the 1/f noise and obtain a rms noise (referred to the preamp input)

$$\sqrt{\overline{v_n^2}} = \sqrt{S_b} \cdot \sqrt{(f_h - f_i)} \approx \sqrt{S_b} \cdot \sqrt{f_h} = 5\mu V$$

and therefore

$$\frac{S}{N} = \frac{V_S}{\sqrt{\overline{v_n^2}}} \leq 0{,}02 << 1$$

Even the highest signal $V_s$ *=100 nV* is **practically invisible on the oscilloscope display**! The noise covers a band *≈ 5 x rms value ≈ 20 µV* and the sinusoidal signal is buried in it!



**Vertical scale 50µV/div**

**Horizontal scale 5µs/div**

# Recovering Narrow-Band Signals

**CASE 1: signal $V_s \leq 100\ nV$ at high frequency $f_s$ = 100 kHz**

**b) observing the power spectrum in FREQUENCY DOMAIN, i.e. on spectrum analyzer display**

**SIGNAL**: the power $P_S = \frac{V_S^2}{2} = 50 \cdot 10^{-16}\ V^2$ is within a bandwidth $\Delta f_S = 1\ Hz$

so that the effective power density of the signal is $\sqrt{S_S} \approx \sqrt{\frac{P_S}{\Delta f_S}} = \frac{70nV}{\sqrt{Hz}}$

**NOISE**: the effective power density at $f_s$ = 100 kHz is $\sqrt{S_b} = 5nV/\sqrt{Hz}$

On the spectrum analyzer display the signal peak is **very well visible above the noise!**

$$\frac{\sqrt{S_S}}{\sqrt{S_b}} = 14 \gg 1$$

**Conclusion**: good S/N can be obtained with a bandpass filter having bandwidth $\Delta f_b$ matched to the signal band $\Delta f_b \approx \Delta f_S$

$$\frac{S}{N} = \sqrt{\frac{P_S}{S_b \Delta f_b}} = \sqrt{\frac{S_S \Delta f_S}{S_b \Delta f_b}} \approx \frac{\sqrt{S_S}}{\sqrt{S_b}} = 14 \gg 1$$

**CASE 2: signal $V_s \leq 100\ nV$ at moderately low frequency $f_s = 1$ kHz**

**a) observing the voltage waveform in the TIME DOMAIN, i.e. on oscilloscope display**

The signal is now at $f_s = 1$ kHz just below the corner frequency $f_c = 2kHz$.
For reducing the 1/f noise we can still use a high-pass filter, but in order to pass the signal the band-limit $f_i$ must be reduced: $f_i \ll f_s = 1$ kHz, typically $f_i = 100$Hz.
The rms noise referred to the input is

$$\sqrt{\overline{v_n^2}} \approx \sqrt{S_b(f_h - f_i) + S_b f_c \ln\left(\frac{f_h}{f_i}\right)} \approx \sqrt{S_b f_h + S_b f_c \ln\left(\frac{f_h}{f_i}\right)} \approx \sqrt{S_b f_h} \approx 5\mu V$$

1/f noise is negligible $\quad S_b f_c \ln\left(\frac{f_h}{f_i}\right) \ll S_b f_h$

and therefore

$$\frac{S}{N} = \frac{V_S}{\sqrt{\overline{v_n^2}}} \leq 0{,}02 << 1$$

The situation is practically equal to that of Case 1: the signal is **practically invisible on the oscilloscope display, it's buried in the noise**!

**CASE 2: signal $V_s \leq 100\ nV$ at moderately low frequency $f_s$ = 1 kHz**

**b) observing the power spectrum in FREQUENCY DOMAIN, i.e. on spectrum analyzer display**

SIGNAL: the power $P_S = \frac{V_S^2}{2} = 50 \cdot 10^{-16}\ V^2$ is within a bandwidth $\Delta f_S = 1\ Hz$

so that the effective power density of the signal is $\sqrt{S_S} \approx \sqrt{\frac{P_S}{\Delta f_S}} = \frac{70nV}{\sqrt{Hz}}$

**NOISE:** due to the **1/f noise**, the effective power density at $f_s$ = 1 kHz is somewhat higher

$$\sqrt{S_n(f_s)} = \sqrt{S_b + S_b \frac{f_c}{f_s}} = \sqrt{S_b}\sqrt{1 + \frac{f_c}{f_s}} = \sqrt{3} \cdot \sqrt{S_b} \approx 8{,}7 \frac{nV}{\sqrt{Hz}}$$

Anyway, on the spectrum analyzer display the signal peak is still **well visible above the noise**

$$\frac{\sqrt{S_S}}{\sqrt{S_n(f_S)}} = 8 > 1$$

**Conclusion**: a bandpass filter with bandwidth $\Delta f_b$ matched to the signal $\Delta f_b \approx \Delta f_s$ still gives a **fairly good S/N**

$$\frac{S}{N} = \sqrt{\frac{P_S}{S_n(f_S)\Delta f_b}} = \sqrt{\frac{S_S \Delta f_S}{S_n(f_S)\Delta f_b}} \approx \frac{\sqrt{S_S}}{\sqrt{S_n(f_S)}} = 8 > 1$$

**CASE 3: signal $V_s \leq 100\ nV$ at low frequency $f_s$ = 10 Hz**

**a) observing the voltage waveform in the TIME DOMAIN, i.e. on oscilloscope display**

The signal is now at $f_s$ = 10 Hz much below the corner frequency $f_c = 2kHz$.
For reducing the the 1/f noise we can still use a high-pass filter, but with strongly reduced band-limit $f_i << f_s$ = 10 Hz, typically $f_i$ = 1 Hz. The rms noise referred to input is

$$\sqrt{\overline{v_n^2}} \approx \sqrt{S_b(f_h - f_i) + S_b f_c \ln\left(\frac{f_h}{f_i}\right)} \approx \sqrt{S_b f_h + S_b f_c \ln\left(\frac{f_h}{f_i}\right)} \approx \sqrt{S_b f_h} \approx 5\mu V$$

1/f noise is negligible $\quad S_b f_c \ln\left(\frac{f_h}{f_i}\right) \ll S_b f_h$

and therefore $$\frac{S}{N} = \frac{V_S}{\sqrt{\overline{v_n^2}}} \leq 0{,}02 << 1$$

The situation is practically equal to that of Case 1 : the signal is **practically invisible on the oscilloscope display, it’s buried in the noise**!

# Recovering Narrow-Band Signals

**CASE 3: signal $V_s \leq 100\ nV$ at low frequency $f_s$ = 10 Hz**

**b) observing the power spectrum in FREQUENCY DOMAIN, i.e. on spectrum analyzer display**

SIGNAL: the power $P_S = \frac{V_S^2}{2} = 50 \cdot 10^{-16}\ V^2$ is within a bandwidth $\Delta f_S = 1\ Hz$

so that the effective power density of the signal is $\sqrt{S_S} \approx \sqrt{\frac{P_S}{\Delta f_S}} = 70 \frac{nV}{\sqrt{Hz}}$

NOISE: due to the **1/f noise**, the effective power density at $f_s$ = 10 Hz is now **much higher**

$$\sqrt{S_n(f_s)} = \sqrt{S_b + S_b \frac{f_c}{f_s}} = \sqrt{S_b}\sqrt{1 + \frac{f_c}{f_s}} = 14{,}2 \cdot \sqrt{S_b} \approx 71 \frac{nV}{\sqrt{Hz}}$$

On the spectrum analyzer display the signal peak is **barely visible, it's equal to the noise**

$$\frac{\sqrt{S_S}}{\sqrt{S_n(f_S)}} \approx 1$$

**Conclusion: the S/N is insufficient** even with a bandpass filter with narrow bandwidth $\Delta f_b$ matched to the signal $\Delta f_b \approx \Delta f_s$

$$\frac{S}{N} = \sqrt{\frac{P_S}{S_b \Delta f_b}} = \sqrt{\frac{S_S \Delta f_S}{S_n(f_S)\Delta f_b}} \approx \frac{\sqrt{S_S}}{\sqrt{S_n(f_S)}} \leq 1$$

# SUMMARY

- For a narrow-band signal plunged in white noise (i.e. with frequency $f_S$ higher than the 1/f noise corner frequency $f_c$ ) a bandpass filter matched to the signal band is very efficient and makes possible to recover signals even so small that they are buried in the wide-band noise.
- For a narrow-band signal plunged in dominant 1/f noise (i.e. with $f_S$ lower than the 1/f noise corner frequency $f_c$) a bandpass filter matched to the signal is still quite efficient and in many cases makes possible to recover the signal. However, if we consider signals at progressively lower frequency $f_S$ , the 1/f noise density at $f_S$ progressively rises, so that the available S/N is progressively reduced.

# OPEN QUESTIONS

- We need **efficient band-pass filters** with very narrow band-width.
  We need to understand how to design and implement such narrow-band filters , but we shall deal with this issue after dealing with the following question.

- If the information is carried by the amplitude of a low-frequency signal, it has to face also 1/f noise. ***It would be advantageous to escape this noise by preliminarly transferring the information to a signal at higher frequency***. However:

  a) how can we transfer the signal to higher frequency?

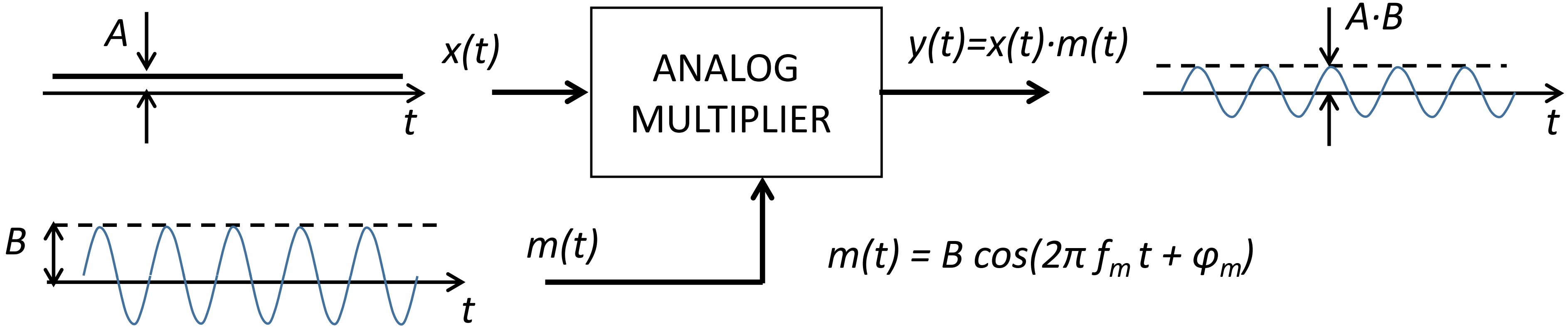
  b) if we transfer to the higher frequency also the 1/f noise that faces the signal, this makes the transfer useless: how can we avoid it?

# OPEN QUESTIONS

- For escaping 1/f noise, a low-frequency signal should be transferred to higher frequency **before it mixes with 1/f noise** of comparable power density: that is, frequency transfer should be done before the stage where the signal meets the 1/f noise source.
- **The frequency-transfer stages have their own nois**e, with different intensity in different types. Unluckily, the types with lowest noise bear other drawbacks, typically a limited capability of transfer, restricted to moderately high frequency.
- For achieving our goal, the **signal must be higher than the noise referred to the input of the frequency-transfer stage**. If with a given stage the signal is not high enough, preamplifying is not advisable because a preamp brings its 1/f noise. In most cases it is better to transfer the signal «as it is» by means of a frequency transfer stage with lower noise and accept the limitations of this stage, typically a moderate operating frequency.

# Moving Signals in Frequency (Signal Modulation)

# Amplitude Modulation with DC signal (ideal)



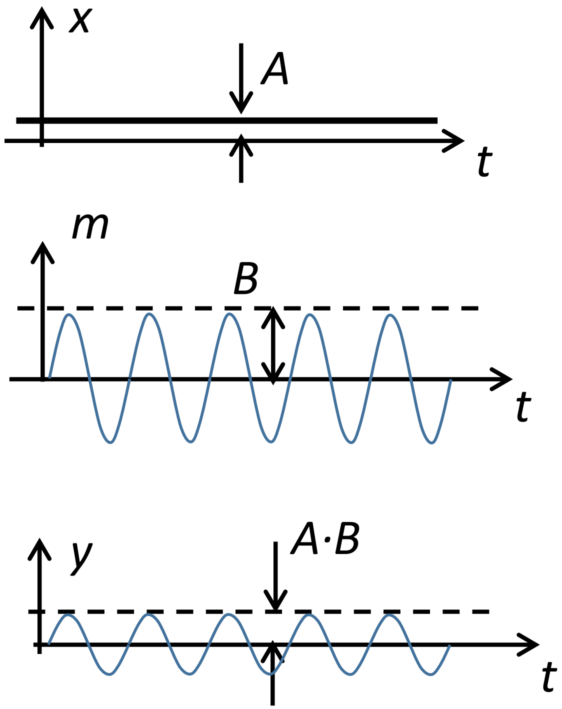

- **Information is brought by** the (VARIABLE) amplitude A of a DC signal $x(t) = A$ .
  (NB: a real DC signal is a signal at very low frequency with very narrow bandwidth)
- An analog multiplier circuit combines the signal with a sinusoidal waveform $m(t)$ *(called reference or carrier)* with frequency $f_m$ and CONSTANT amplitude $B$
- The information is transferred to the amplitude of a sinusoidal signal $y(t)$ at frequency $f_m$

$$y(t) = A \cdot B \cos(2\pi f_m t + \varphi_m)$$

# Amplitude Modulation with DC signal (ideal)



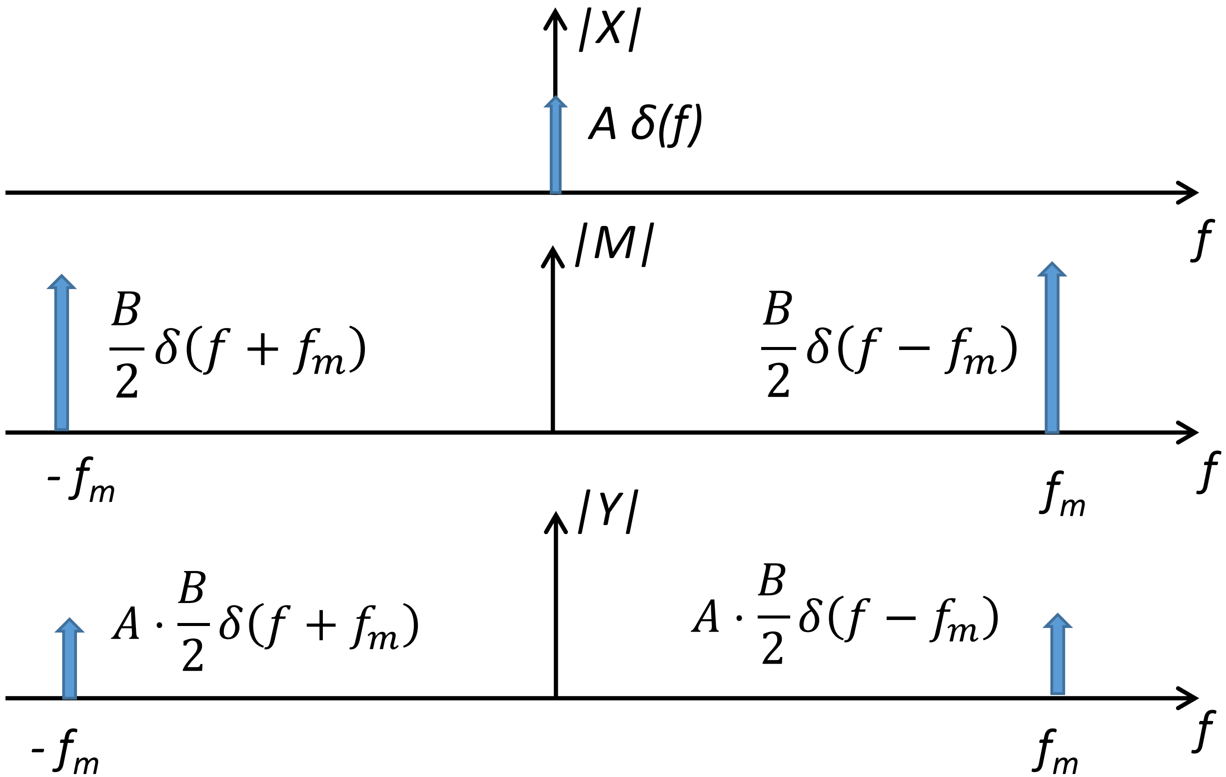
## TIME DOMAIN




$y(t) = x(t) \cdot m(t) =$

$= A \cdot B \cos(2\pi f_m t + \varphi_m)$

## FREQUENCY DOMAIN




$Y(f) = X(f) * M(f)$

The signal is shifted in frequency by $+f_m$ and $-f_m$ and in phase by $+\varphi_m$ and $-\varphi_m$ respectively

# Convolution in the Frequency Domain



In the **time domain (TD)** the amplitude modulation is the **multiplication** of the signal *x(t)* (with variable amplitude A ) by the reference waveform *m(t)* (with standard amplitude B)

$$y(t) = x(t) \cdot m(t)$$

In the **frequency domain (FD)** it is the **convolution** of the transformed signal *X(f)* by the transformed reference M*(f)*

$$Y(f) = X(f) * M(f) = \int_{-\infty}^{\infty} X(\alpha)\, M(f-\alpha) d\alpha$$

Convolution is more complicated in FD than in TD because:

1. the functions to be convolved are twofold, that is, they run in the positive and negative sense of the frequency axis
2. **Complex** values must be summed at every frequency for obtaining *Y(f)*.

**In general** the result of FD convolution is not as intuitive as that of TD convolution and the module |Y(f)|is **NOT** given by the convolution of |X(f)| and|M(f)|

$$|Y(f)| \neq |X(f)| * |M(f)|$$

we must first compute the real and imaginary parts of Y(f) and then obtain |Y(f)|

# Amplitude Modulation with Narrow-Band Signal



**In the cases here considered**, however, the issue is remarkably simplified because

a) *X(f)* is confined in a **narrow** bandwidth $\Delta f_s$

b) *M(f)* has a line spectrum with (fundamental) frequency $f_m$ that is much greater than the signal bandwidth $\boldsymbol{f_m >> \Delta f_s}$

In the convolution *X(f) * M(f)* each line of M(f) acts on *X(f)* as follows

- Shifts in frequency every component of *X(f)* by + $\boldsymbol{f_m}$ and $\boldsymbol{-f_m}$
  (i.e. adds to each frequency + $\boldsymbol{f_m}$ and $\boldsymbol{-f_m}$ )
- Shifts in phase every component of *X(f)* by $+\varphi_m$ and $-\varphi_m$
  (i.e. adds to every phase $+\varphi_m$ and $-\varphi_m$ )

*In cases with $\Delta f_s << \boldsymbol{f_m}$, there is **no sum of complex numbers** to be computed because at any frequency f there is at most one term to be considered, all other terms are negligible.*

The result of the convolution is easily visualized: every line of *M(f)* shifts *X(f)* in frequency and adds to *X(f)* its phase. Therefore, |Y(f)|is well approximated by the convolution of |X(f)| and|M(f)| and $|Y(f)|^2$ by the convolution of $|X(f)|^2$ and$|M(f)|^2$

$$\boxed{|Y(f)| \cong |X(f)| * |M(f)|}$$

$$\boxed{|Y(f)|^2 \cong |X(f)|^2 * |M(f)|^2}$$

# Amplitude Modulation with Narrow-Band Signal



**Example of quasi-DC**

NB signal (with very long $T_S$)

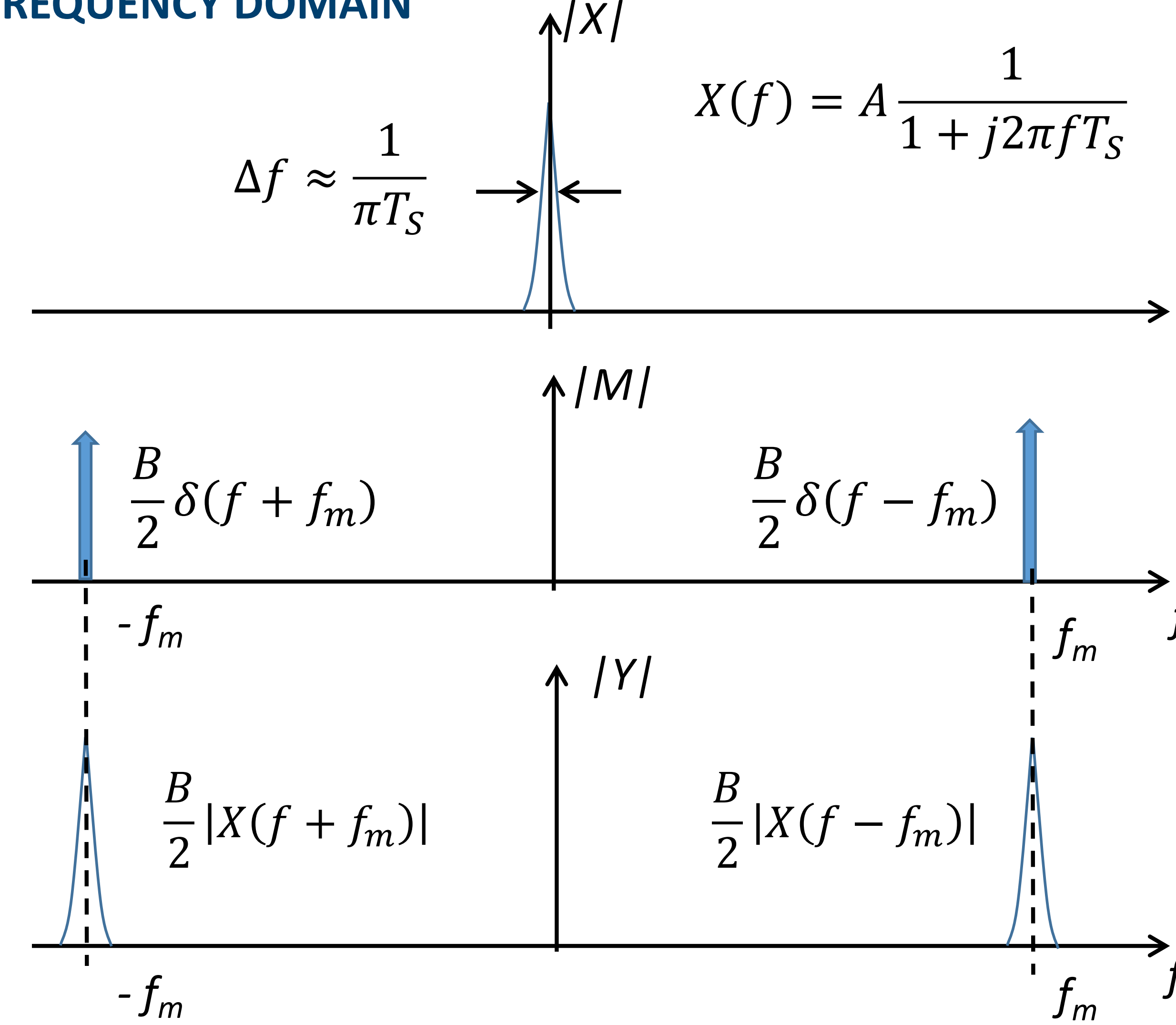
$$x(t) = 1(t) \cdot \frac{A}{T_S} e^{-\frac{t}{T_S}}$$

$m(t) = B \cos(2\pi f_m t + \varphi_m)$

with $f_m >> 1/T_S$

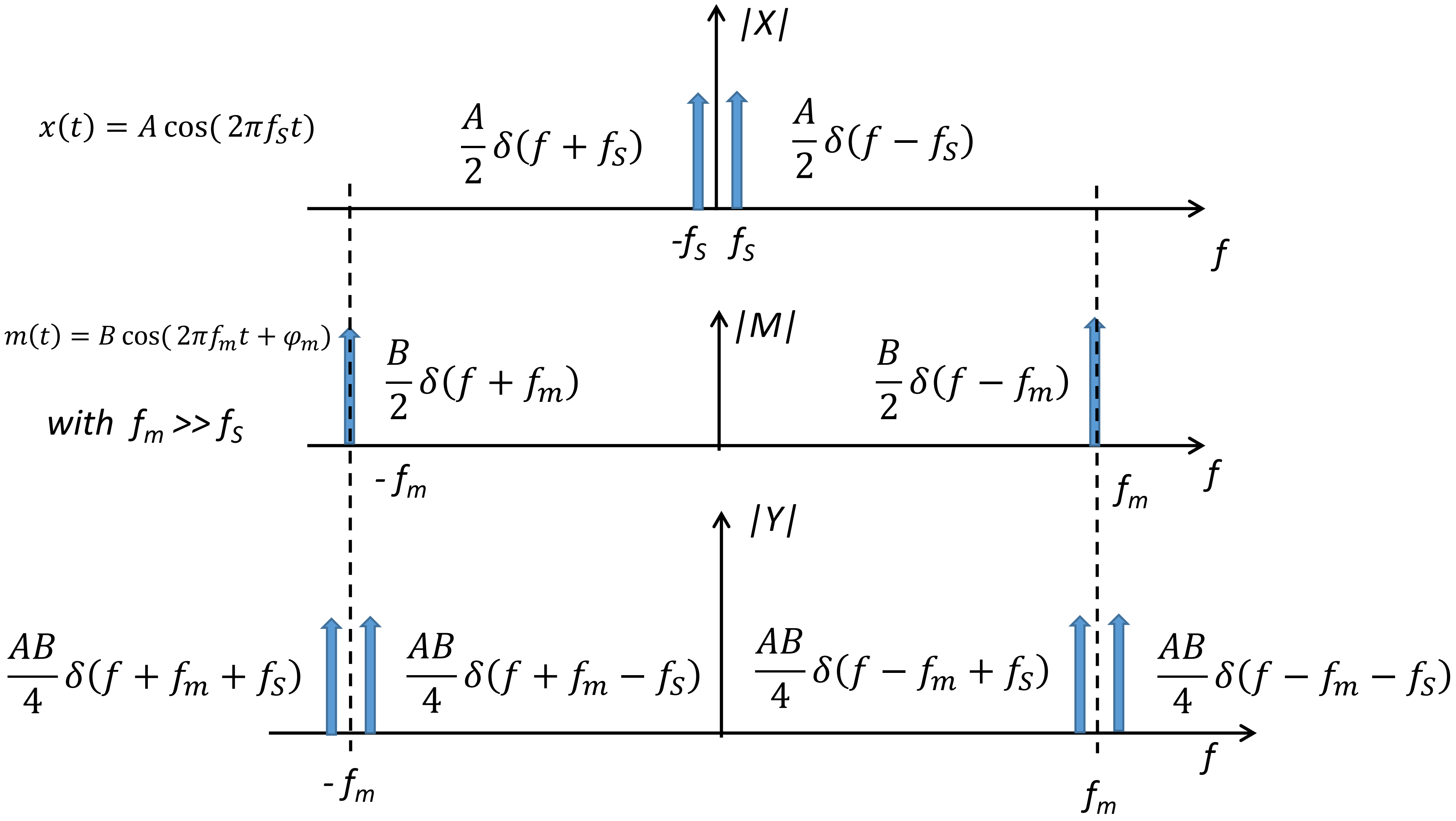
$y(t)=x(t)\cdot m(t)$

**FREQUENCY DOMAIN**




$Y(f) = X(f) * M(f)$

# Amplitude Modulation with Sinusoidal Signal



# Amplitude Modulation with sinusoidal signal



By exploiting the a well known trigonometric equation

$$\cos\alpha\cos\beta = \frac{1}{2}\cos(\alpha-\beta) + \frac{1}{2}\cos(\alpha+\beta)$$

in cases with sinusoidal signal and sinusoidal reference

$$x(t) = A\cos(\,2\pi f_S t)$$

$$m(t) = B\cos(\,2\pi f_m t + \varphi_m)$$

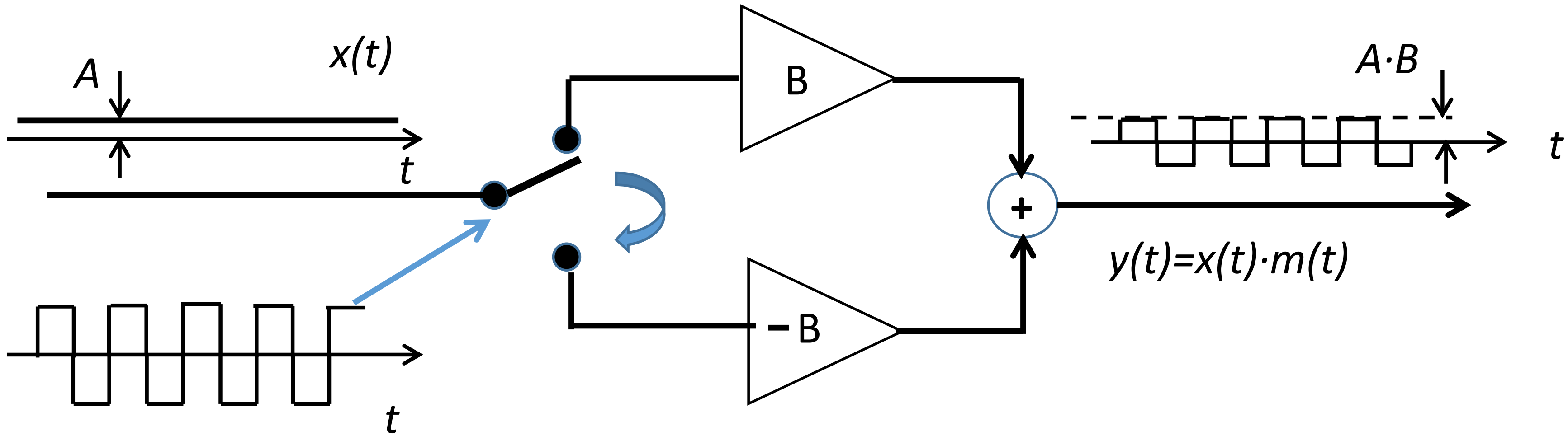
the result is directly obtained

$$\boxed{y(t) = x(t)\cdot m(t) = \frac{AB}{2}\cos[2\pi(f_S - f_m)t - \varphi_m] + \frac{AB}{2}\cos[2\pi(f_S + f_m)t + \varphi_m]}$$

# Squarewave Amplitude Modulation



Modulation with a squarewave reference *m(t)* can be implemented with circuits based simply on switches and amplifiers, without analog multipliers

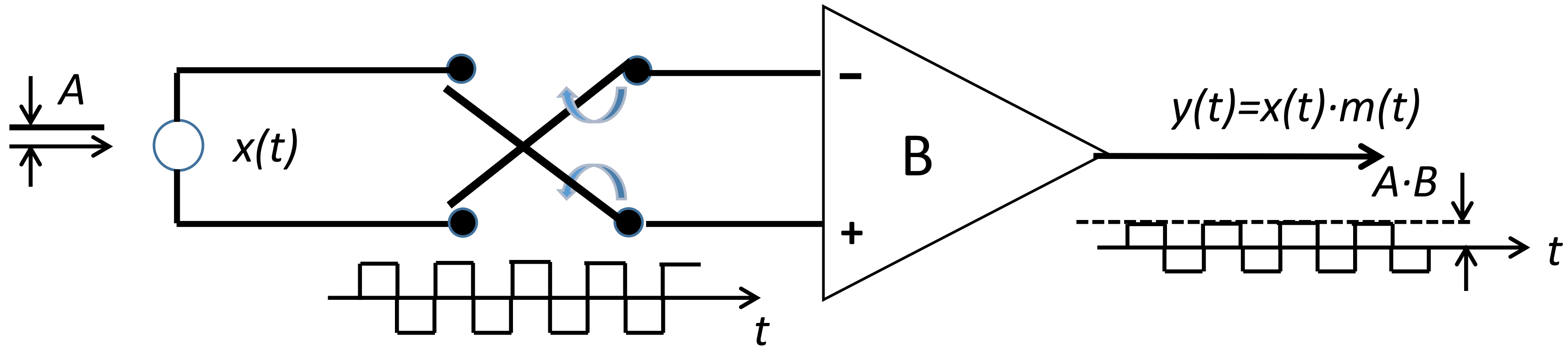



- In such cases, the circuit noise referred to the input is due mostly to the switch-devices and is much lower than that of analog multiplier circuits.
- **Metal-contact switches** have the lowest noise, but they can operate at limited switching frequency, typically up to a few 100Hz
- **Electronic switches** (MOSFET, diodes, etc.) operate up to very high frequencies but have fairly higher noise (anyway MOSFETs operating as switch-device have lower noise than MOSFETs operating as amplifier devices).

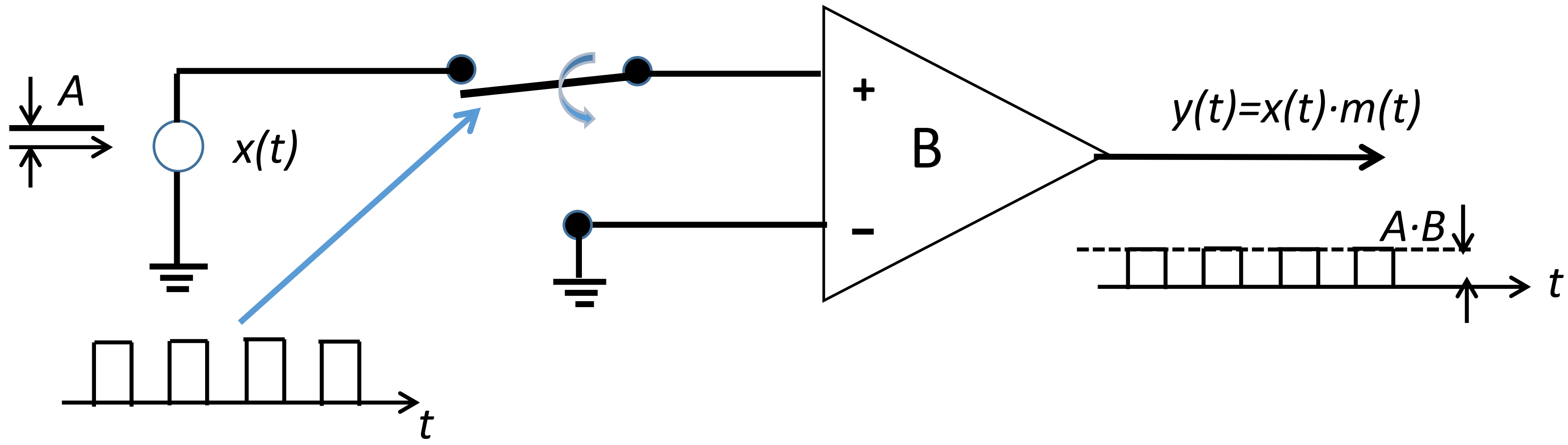
# Squarewave Amplitude Modulation



Switching example: differential amplifier with alternated input polarity

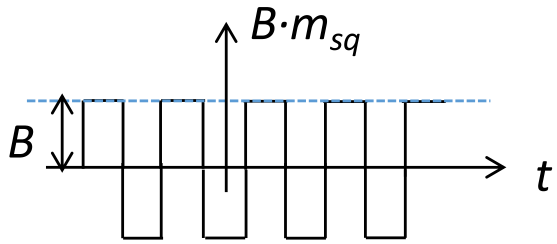



Switching example: chopper (ON-OFF modulation)

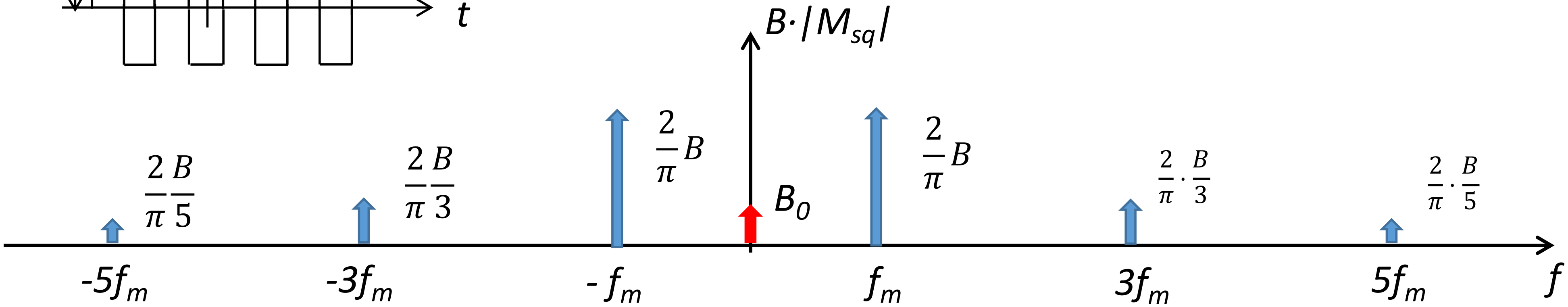


# Squarewaves and F-transforms






$m_{sq}(t)$ = symmetrical squarewave (from +1 to-1) at frequency $f_m$



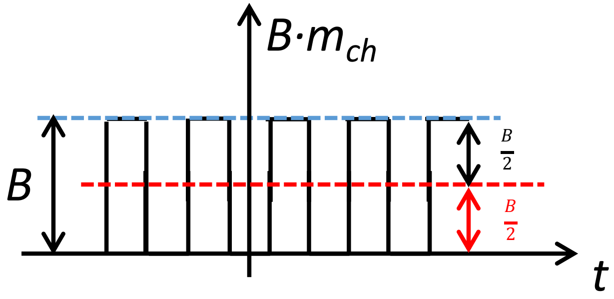

$$\boxed{B \cdot M_{sq}(f) = B \cdot \sum_{k=0}^{\infty} \frac{b_{2k+1}}{2} \left[\delta(f - f_{2k+1}) + \delta(f + f_{2k+1})\right]}$$

with $\begin{cases} f_{2k+1} = (2k+1) f_m \\ b_{2k+1} = \dfrac{(-1)^k}{(2k+1)} \cdot \dfrac{4}{\pi} \end{cases}$

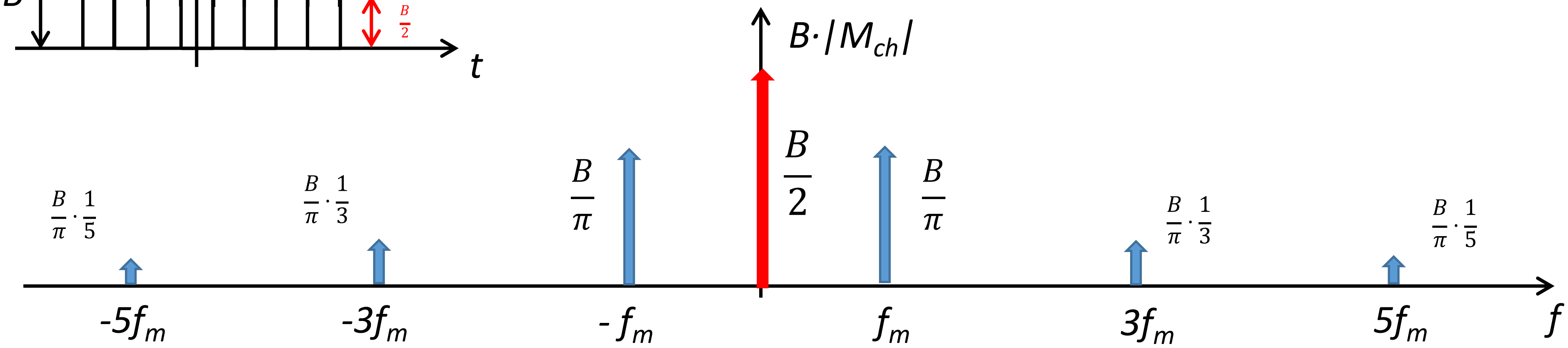
In the amplitude modulation:

- each line of the reference M acts like a simple sinusoidal reference, i.e. shifts by its frequency and its phase the signal X and multiplies it by the amplitude $B \cdot b_{2k+1}$
- if the squarewave is not perfectly symmetrical (e.g. it has asymmetrical amplitude and/or duration of positive and negative parts) there is also a finite DC component with amplitude $B_0$ (possibly very small)
- the DC component does NOT transfer the signal X in frequency, just «amplifies» it by $B_0$

# Squarewaves and F-transforms




$m_{ch}(t)$ = chopper squarewave (from +1 to zero) at frequency $f_m$




Chopper squarewave with amplitude B =

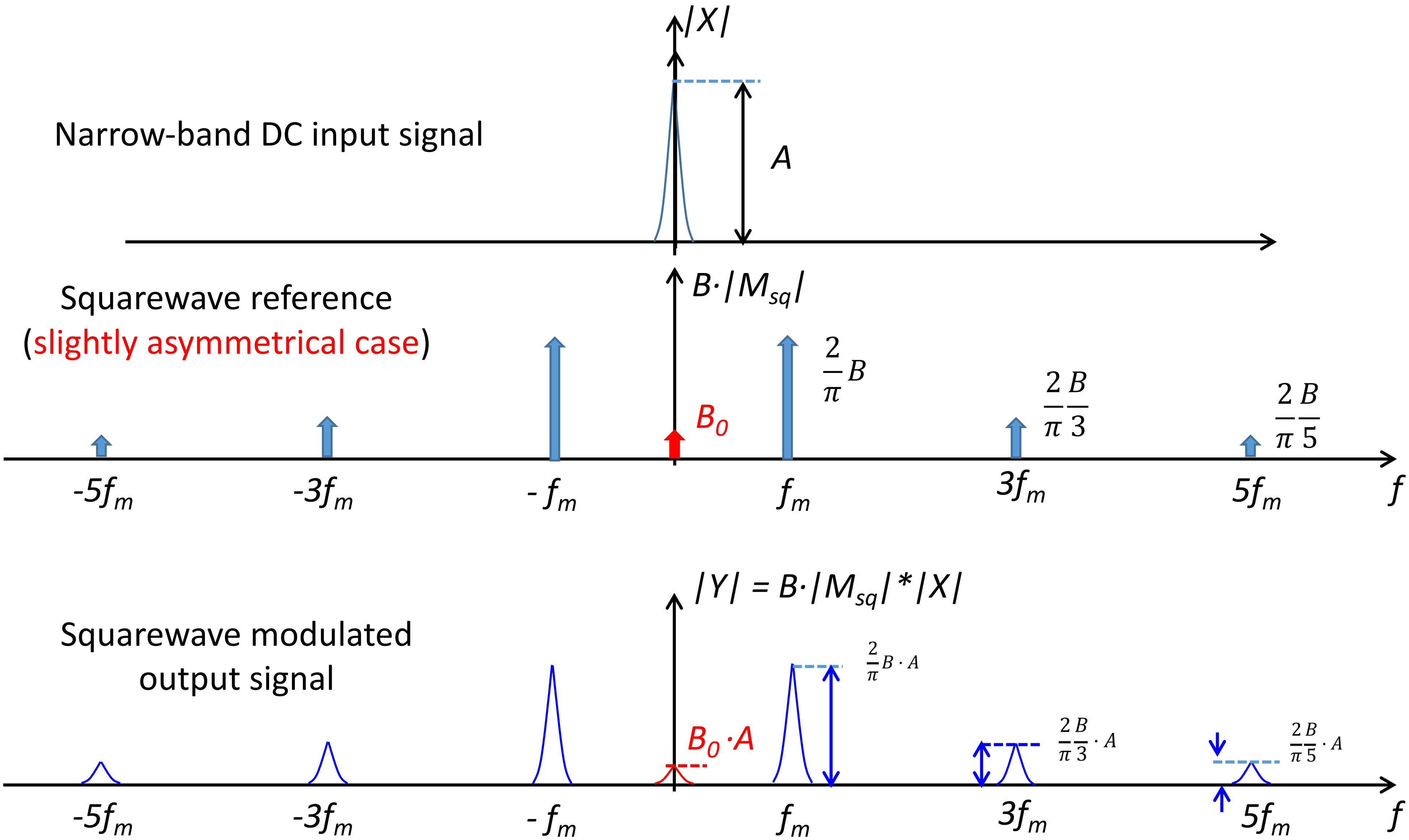
= Symmetrical squarewave with amplitude B/2 + DC component with amplitude B/2

$$\boxed{B \cdot M_{ch}(f) = \frac{B}{2} \cdot M_{sq}(f) + \frac{B}{2} \cdot \delta(f)}$$

In the amplitude modulation by a chopper:

- a replica of the signal X «amplified» by B/2 is transferred in frequency by the squarewave
- another replica of X «amplified» by B/2 is NOT transferred, it stays where it is

# Squarewave Amplitude Modulation



# Summary and Prospect



- As intuitive, narrow-band filtering is very effective for recovering narrow-band signals immersed in wide-band noise
- Besides wide-band noise, however, other components with power density increasing as the inverse frequency (1/f noise) are ubiquitous in electronic circuitry (amplifiers etc.). In the low-frequency range they are indeed dominant.
- At low frequencies the 1/f noise added by the circuitry is overwhelming, so that the solution of narrow-band filtering becomes progressively less effective and finally insufficient for recovering signals with progressively lower frequency
- An effective approach to recover a low-frequency signal is to move it to higher frequency before the addition of 1/f noise. That is, to modulate the signal before the circuitry that contains the 1/f noise sources
- Narrow-band filtering can then be employed to recover the modulated signal; we will now proceed to analyze methods and circuits for narrow band filtering