Principles of Communications Labs

EE311L

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In this set of laboratory experiments, the students will display and measure signals in both domains: time and frequency. In order to do so, we will use a quite modern piece of equipment: a digital storage oscilloscope (GW-Instek GDS1042). This oscilloscope is able not only to display waveforms in the time domain, but it can also store samples of the waveform in its internal memory and it is also able to compute and display the modulus of the Fourier transform (via the computation of the FFT). We cannot present the function of this equipment without introducing some basic definitions and appropriate terminology.

I. rms value of a signal:

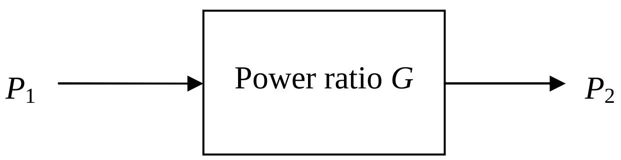
Consider the periodic signal *x*(*t*). Its rms value is defined as:

$$\mathrm{X_{rms}} = \sqrt{\frac{1}{\mathrm{T}}\int_0^{\mathrm{T}} x^2(t)dt} \qquad (1)$$

where T is the fundamental period of the signal. In the case of a sine wave, $x(t) = \mathrm{A}\cos(\omega t + \varphi)$, $\mathrm{X_{rms}} = \mathrm{A}/\sqrt{2}$, i.e. the peak amplitude divided by $\sqrt{2}$. However, if the signal is not sinusoidal, the above relation won't be true and you should use the defining formula (1). Consider for example a square wave oscillating between +A and –A. Its rms value is A. So, when you are measuring ac voltages, unless the signal is sinusoidal, you should make sure that your voltmeter is a true rms voltmeter, otherwise its indication will be meaningless. In many instruments such as spectrum analyzers, the amplitude of each individual sinusoidal wave composing the signal is given in rms or even in decibels (*dB*).

II. The decibel

In electrical engineering, we are used to measure gain of stages using power ratios. Consider the following system:



The average input power is P_1 and the average output power is P_2. The gain of the system is given by:

$$G = \frac{P_2}{P_1}$$

If we consider n such systems in cascade, with gains G_1, G_2, …, G_n, the overall gain is given by the product:

$$G = G_1 G_2 ... G_n$$

In many communication systems, values can vary from about 10^{-10} up to 10^{10}. The enormous range cannot be represented using a linear scale. In order to have a more sensible way to express this, we can use a logarithmic scale. We define the Bel (*B*) (in honor of Alexander Graham Bell, the inventor of the telephone) as:

$$G_{Bel} = \log_{10} G = \log_{10} \frac{P_2}{P_1} \quad (2)$$

Using Bels, the previous range becomes –10 Bel to 10 Bel. Furthermore, the gain of the cascade of n systems is now given by the addition of the individual gains expressed in Bels. The problem with the Bel as a unit is that it is too coarse. An increase of 1 Bel implies that the power ratio has been multiplied by 10. This is why it is more practical to use the decibel (*dB*):

$$G_{dB} = 10 \log_{10} G \quad (3)$$

With this unit, doubling the power ratio amounts to adding about 3dB ($10\log_{10}2$) and the above range becomes – 100 dB to 100 dB.

Sometimes, the decibel is used to specify a voltage ratio. If the impedance level is the same at the two ports: input and output, then, since $P_i = \frac{V_i^2}{R}$, we obtain the following relation:

$$G_{dB} = 20 \log_{10} \frac{V_2}{V_1} \quad (4)$$

In this case, the gain indicates a voltage gain. In many instruments, the amplitude scale is given in *dB*. At that time, the value indicated is not anymore a unit-less quantity such as a power or a voltage gain but it represents a power or a voltage expressed in *dB*. In this case, we use the notation *dB* along with a suffix: *dBm*, *dBW* or *dBV*.

If a power *P* is expressed in watts (*W*), then it can also be expressed in *dBW* as:

$$P_{dBW} = 10 \log_{10} P_W \quad (5)$$

If it is expressed in milliWatts (*mW*), it can be expressed in *dBm* as

$$P_{dBm} = 10 \log P_{mW} \quad (6)$$

If we measure an rms voltage *V* in volts (*V*), we can express it in dBV as:

$$V_{dBV} = 20 \log_{10} V_V \quad (7)$$

Of course, the inversion is provided by

$$V_V = 10^{\frac{V_{dBV}}{20}} \quad \text{or} \quad P_{mW} = 10^{\frac{P_{dBm}}{10}} \quad \text{or} \quad P_W = 10^{\frac{P_{dBW}}{10}} \quad (8)$$

The following table (taken from: Richard Clay and Tewfik Doumi, Communication EE411 lab manual, INELEC, 1982) can be used to solve most problems expressed in dB.

Decibels	Power ratio	Voltage ratio
1	1.26	1.12
2	1.58	1.26
3	2.00	1.41
4	2.51	1.58
5	3.16	1.78
6	4.00	2.00
7	5.01	2.24
8	6.31	2.51
9	8.00	2.82
10	10.00	3.16

For example: 65 *dB* = (60 + 5) *dB*

Power ratio = $10^6 \times 3.16 = 3160\,000$

Voltage ratio = $10^3 \times 1.78 = 1780$

– 23 *dB* = – (20 + 3) *dB*

Power ratio = $10^{-2} \div 2.00 = 0.005$

Voltage ratio = $10^{-1} \div 1.41 = 0.0707$

III. Sampling Signals and displaying Spectra using FFT

If we sample a bandlimited waveform, we do not lose any information if the sampling rate is high enough.

The Sampling Theorem:

Given a bandlimited signal x(t) with spectrum X(f) = 0 for |f | > W. The signal can be recovered from its samples $x(nT_s)$ taken at a rate $f_s = 1/T_s$ with $f_s \geq 2W$.

$$x(t) = \sum_{n=-\infty}^{+\infty} x(nT_s)\,\mathrm{sinc}\left(\frac{t - nT_s}{T_s}\right)$$

The function sinc is defined by:

$$x(t) = \mathrm{sinc}\, t = \frac{\sin \pi t}{\pi t}$$

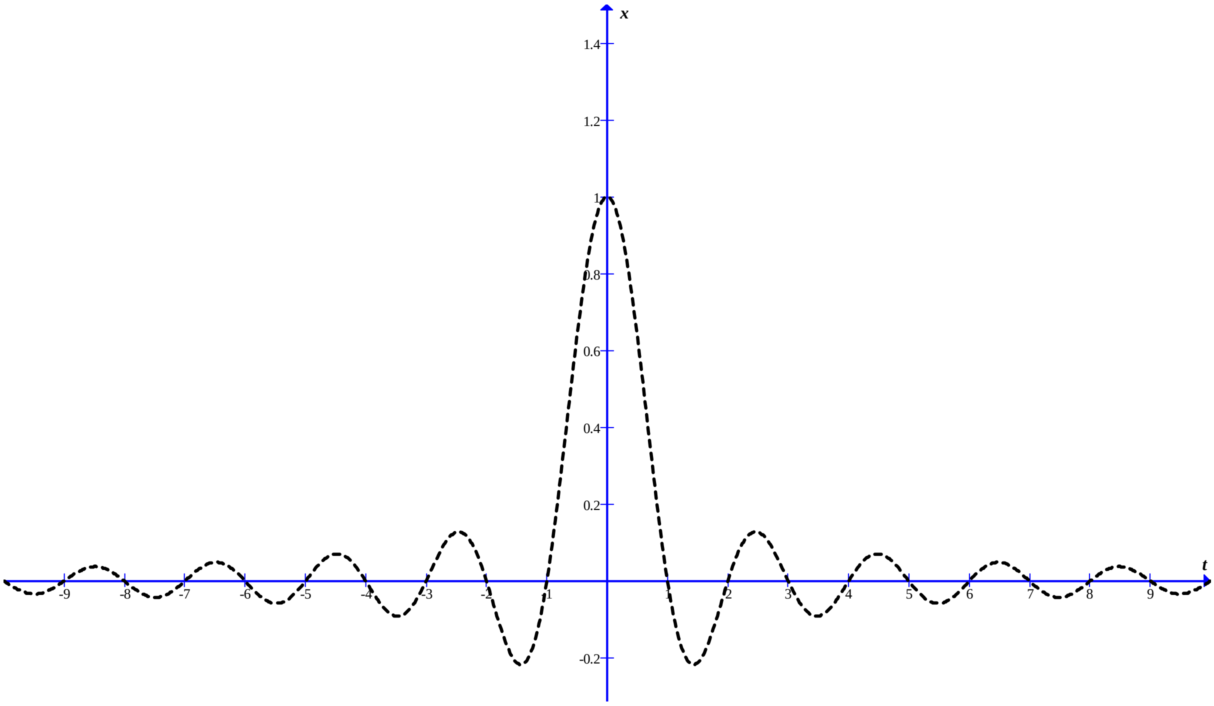


Figure-1 The Sinc function

The obtained sequence $x_1(n) = x(nT_s)$ has a "digital" spectrum defined as:

$$X_1(\omega) = \sum_{n=-\infty}^{+\infty} x_1(n) e^{-jn\omega}$$

along with

$$x_1(n) = \frac{1}{2\pi} \int_{-\pi}^{+\pi} X_1(\omega) e^{jn\omega} d\omega$$

We can obtain a relationship between spectra $X(f)$ and $X_1(\omega)$ by noticing that:

$$x(t) = \int_{-\infty}^{+\infty} X(f) e^{j2\pi ft} df$$

We can limit the above integral to the interval $[-f_s/2, +f_s/2]$ since the spectrum is zero above this frequency.

$$x(t) = \int_{-\frac{f_s}{2}}^{\frac{f_s}{2}} X(f) e^{j2\pi ft} df$$

so $$x(nT_s) = \int_{-\frac{f_s}{2}}^{\frac{f_s}{2}} X(f) e^{j2\pi fnT_s} df \qquad (9)$$

but we have: $$x(nT_s) = x_1(n) = \frac{1}{2\pi} \int_{-\pi}^{\pi} X_1(\omega) e^{jn\omega} d\omega$$

Replacing $2\pi fT_s$ by ω in (9), we obtain:

$$x(nT_s) = \frac{1}{2\pi T_s} \int_{-\pi}^{\pi} X\left(\frac{\omega}{2\pi T_s}\right) e^{jn\omega} d\omega$$

Finally: $$X(f) = T_s X_1(2\pi T_s f) \text{ or } X_1(\omega) = \frac{1}{T_s} X\left(\frac{\omega}{2\pi T_s}\right)$$

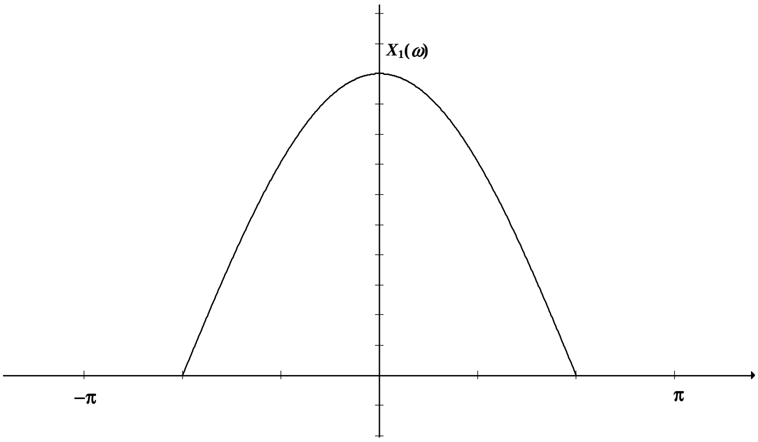


Figure -2 Spectrum corresponding to the sequence

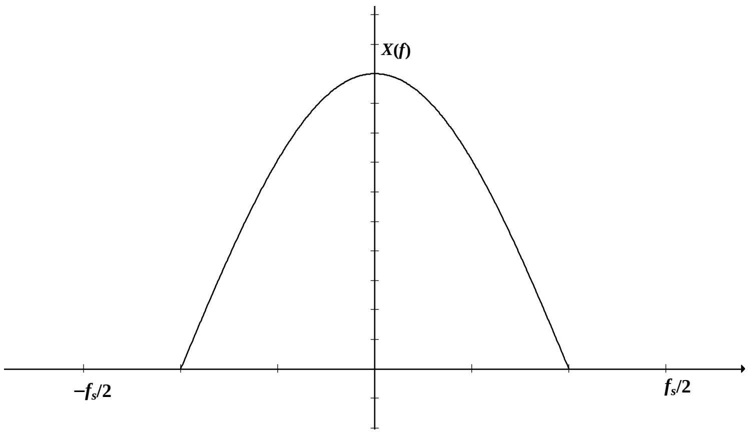


Figure -3 Spectrum corresponding to the signal

The above two figures show clearly the relationship between the two spectra. The two spectra are identical except for a gain factor T_s and a frequency scale where $\omega = \pi$ in figure 2 corresponds to $f = f_s/2$ in figure 3. This means that if a signal is bandlimited, then we can compute its spectrum by computing the one of the sequence obtained by sampling it.

We can also remark that the spectra are computed using an infinite duration sequence. In practice, we have only access to a finite realization of the signal. This is equivalent to multiply the original signal by a finite duration "window".

So, we obtain $y(t) = x(t)w(t)$. In the frequency domain, we have: $Y(f) = X(f) * W(f)$. $w(t)$ is the window function. In terms of samples, the duration T of the window is $T = NT_s$. The spectrum of the obtained sequence is:

$$Y_1(\omega) = \sum_{n=0}^{N-1} y_1(n)e^{-jn\omega}$$

The Discrete Fourier Transform (DFT) corresponds to sampling the above spectrum in the frequency domain: $\omega_k = \dfrac{2\pi k}{N}, k = 0, \cdots, N-1$. The DFT can be computed very efficiently using a powerful algorithm: The Fast Fourier Transform (FFT). The DFT of the sequence $y_1(n)$ is given by:

$$Y_1(k) = Y_1\left(\frac{2\pi k}{N}\right) = \sum_{n=0}^{N-1} y_1(n) e^{-j\frac{2\pi nk}{N}} \qquad ; k = 0, \cdots, N-1$$

These samples are related to the analog spectrum by:

$$Y_1(k) = \frac{1}{T_s} Y\left(\frac{k}{NT_s}\right) = f_s Y\left(\frac{kf_s}{N}\right)$$

The GDS1042 displays $20\log_{10}|Y_1(k)|$ properly scaled for a frequency range $f \in [0, f_s/4]$ or for $k = 0, \cdots, N/4 - 1$ frequency samples.

Effect of the window

Since the displayed spectrum is the one of the windowed signal *y*(*t*) and not the true signal *x*(*t*), the choice of the window becomes primordial.

Assume we generate a sine wave $x(t) = A\cos(2\pi f_0 t)$. Its spectrum is $X(f) = \frac{A}{2}\delta(f - f_0) + \frac{A}{2}\delta(f + f_0)$. If we use a rectangular window, $w(t) = \Pi\left(\frac{t}{T}\right)$, the spectrum becomes $Y(f) = X(f) * W(f)$, $W(f) = T\,\text{sinc}\, fT$ .So, the displayed spectrum will be the one of the following function:

$$Y(f) = \frac{AT}{2}\text{sinc}\,T(f - f_0) + \frac{AT}{2}\text{sinc}\,T(f + f_0)$$

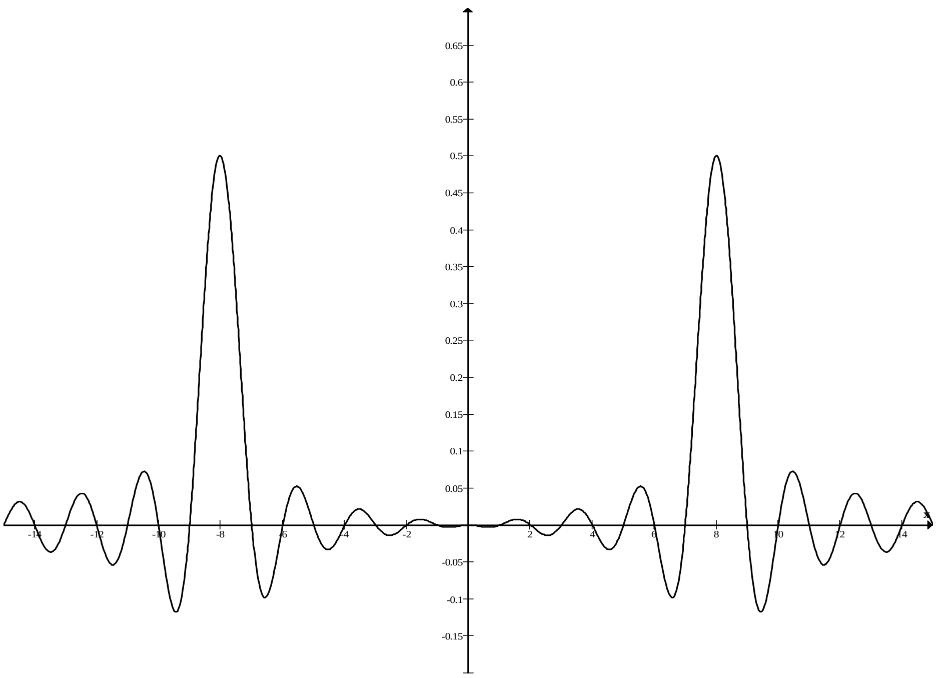


Figure -4 Spectrum of *y*(*t*)

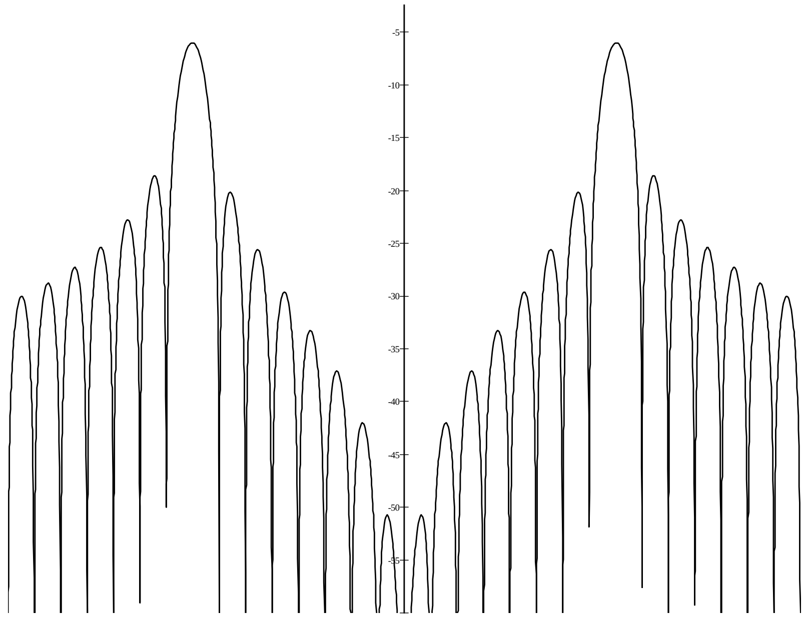


Figure -5 Power Spectrum of *y*(*t*) in dB

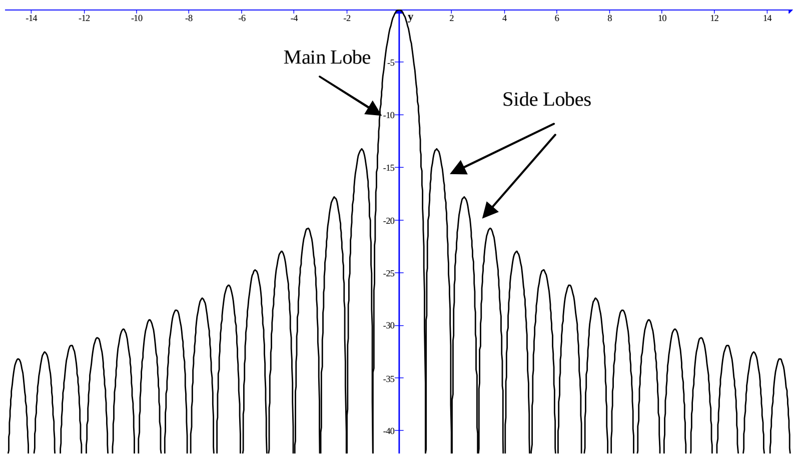


Figure 6 Fourier Transform of a Rectangular Window in dB

We remark that the window has two main effects. The first one is an uncertainty on the location of the frequency of the carrier. This is due to the width of the main lobe. In our example, its width is 2 Hz. This means that if two signals differ in frequency by less than the width of the main lobe, it will be impossible to distinguish one from the other. The other effect is the leakage to other frequencies due to the height of the side lobes. The side lobes can cover a weak signal and make it impossible to be discernable. Furthermore, their value will add to the true one. This implies that the amplitudes will be in error. In order to make precise measurements, we should use a window that has a very narrow main lobe and very low side lobes. However, it happens that these two objectives are antagonistic. We can optimize one of them, but only at the expense of the other. The windows that have low side lobes have a wider main lobe. In this case, we can increase the frequency precision by increasing the sampling rate. One of the simplest windows is the cosine window: $w(t) = \cos\frac{\pi t}{T}\Pi(\frac{t}{T})$. Its Fourier transform is: $W(f) = \frac{2T}{\pi}\frac{\cos\pi fT}{1-4f^2T^2}$.

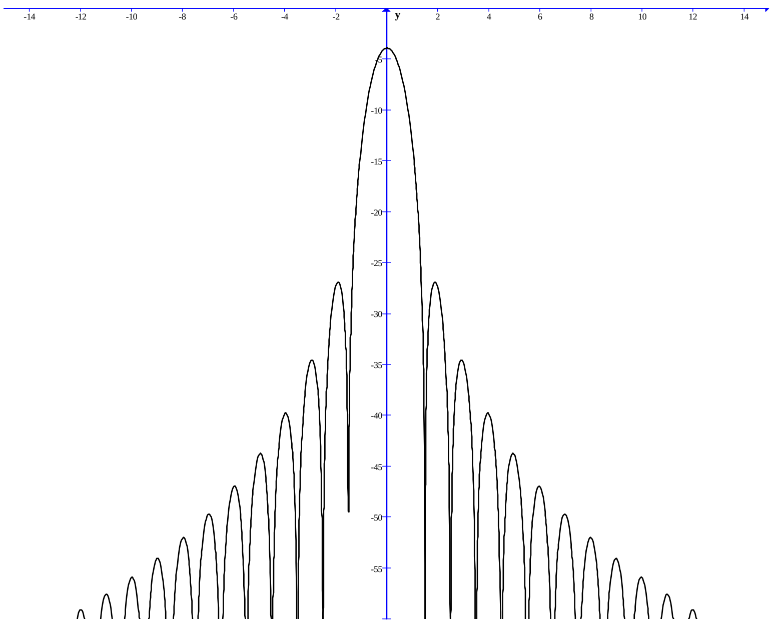


Figure 7 Fourier Transform of a Cosine Window in dB

We can remark that the side lobes are much smaller. However, the main lobe width is twice the one of the rectangular window.

A secondary effect of the window is an amplitude scaling.
For the sinewave multiplied by a rectangular window, the displayed spectrum will be $Y(f)$ for frequencies between zero and $f_s/2$.

$$Y(f) = \frac{AT}{2}\text{sinc}\left[T\left(f - f_0\right)\right] \quad ; \quad 0 \le f \le f_s/2$$

The DFT is: $$Y_1(k) = \frac{ATf_s}{2}\text{sinc}\left[T\left(\frac{kf_s}{N} - f_0\right)\right] \quad ; \quad 0 \le k \le \frac{N}{2} - 1$$

The duration of the window is $T = NT_s$. The computed spectrum is:

$$Y_1(k) = \frac{AN}{2}\text{sinc}\left[k - \frac{Nf_0}{f_s}\right]$$

If we consider now the cosine window, the above results become:

$$Y(f) = \frac{AT}{\pi} \frac{\cos\left[\pi T\left(f - f_0\right)\right]}{1 - 4T^2\left(f - f_0\right)^2}$$

$$Y_1(k) = \frac{AN}{\pi} \frac{\cos\left[\pi\left(k - \frac{Nf_0}{f_s}\right)\right]}{1 - 4\left(k - \frac{Nf_0}{f_s}\right)^2}$$

A very popular function is the Hanning function (named after Julius Von Hann):

$$w(t) = 0.5\left(1 + \cos\frac{2\pi t}{T}\right)\Pi\left(\frac{t}{T}\right)$$

with transform:

$$W(f) = \frac{T}{2} \frac{\operatorname{sinc} fT}{\left(1 - f^2 T^2\right)}$$

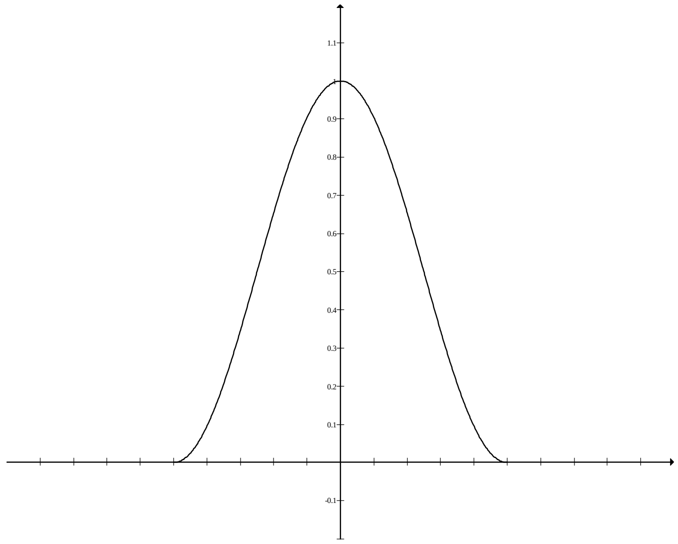


Figure 8 Hanning window

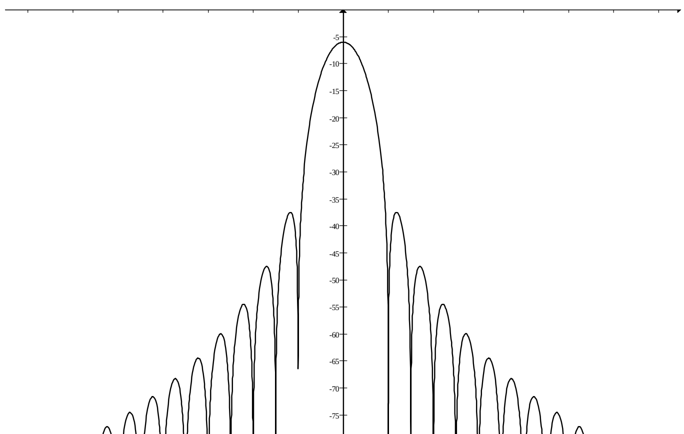


Figure 9 Fourier Transform of the Hanning Window in dB

The Hanning window has much smaller side lobes and a wider main lobe (three times the one of the rectangular window). With the Hanning window, the previous result becomes:

$$Y(f) = \frac{AT}{4}\frac{\text{sinc}\left[T\left(f - f_0\right)\right]}{1 - T^2\left(f - f_0\right)^2}$$

$$Y_1(k) = \frac{AN}{4}\frac{\text{sinc}\left[\left(k - \frac{Nf_0}{f_s}\right)\right]}{1 - \left(k - \frac{Nf_0}{f_s}\right)^2}$$

With the Hanning window, the scale factor is $N/2$. It is $2N/\pi$ for the cosine window.

Other windows are the Hamming window, the Flat Top window, the Blackman window, the Kaiser window, etc. The digital oscilloscope GDS1042 uses four different windows: rectangular, Hanning, Blackman and Flat Top windows.

Lab 1

FFT and Windows

Purpose

The objectives of this experiment are to learn how to generate and analyze waveform on a computer, to see the effect of a window on the spectral analysis of a waveform and to observe the effect of aliasing in we select a too small sampling frequency.

MATLAB

In this first experiment, we are going to generate and analyze waveforms using a specific programming environment: MATLAB. MATLAB (MATrix LABoratory) is a programming language dedicated essentially to numerical computations. It is an interpretive language, meaning that the statements are translated and executed one after the other. This is quite different from the C language that you have learned in your programming courses. The basic data structure in MATLAB is the two dimensional array of complex numbers. It accepts also row and column vectors. Arrays are introduced row by row. For example:

```
>> A = [1 2 3; 4 5 6]
A =
    1   2   3
    4   5   6
```

The above statement has generated a 2×3 matrix. We remark also that the result of the statement is displayed right after (and below) the assignment statement. If we want to avoid this behavior, we should end the statement by a semicolon (;). We can generate row vectors that have elements which are equally spaced by using the colon (:) operator.

```
>> T = (0 : 0.5 : 2)
T =
      0   0.5000   1.0000   1.5000   2.0000
```

The above statement generates a five element row vector starting at 0, ending at 2 with a step of 0.5 between values. If the middle number is omitted, the increment will always take the default value 1. We can also transpose vectors and matrices using the apostrophe operator.

```
>> B = A'
B =
    1   4
    2   5
    3   6
```

Furthermore, MATLAB statements operate on whole arrays. This means that we do not require iterations to perform matrix operations. For example:

```
>> C=A*B
C =
    14    32
    32    77
```

The 2×2 C matrix is created and is equal to the product of the 2 matrices A and B. The elements of the matrix C are given by $c_{ij} = \sum_{n=1}^{N} a_{in} b_{nj}$ where N is the number of columns of A and the number of rows of B. When using this type of statements, we must make sure that the matrix product is possible. There exist also array operations. They are characterized by the dot preceding the operator.

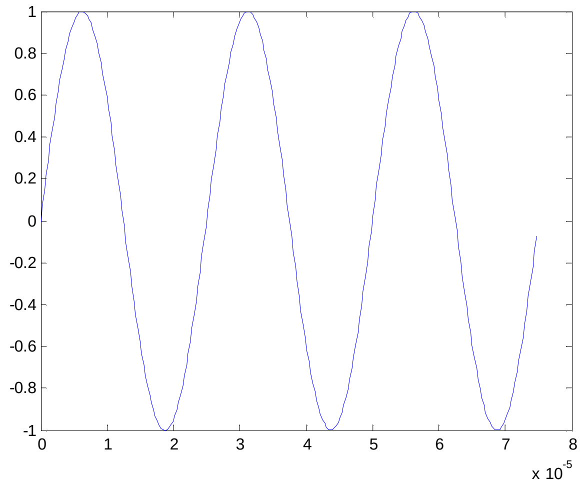
```
>> D= A .*A
D =
     1     4     9
    16    25    36
```

In this case the 2 matrices must have the same dimensions and the elements of D = A.*B are computed using $d_{ij} = a_{ij} b_{ij}$.

Example of a short MATLAB program:

The following short program generates L = 256 samples representing 3 periods of a 40 kHz sinewave. The first step is to determine the sampling period. The period of the sinewave is $T_0 = \frac{1}{f_0} = \frac{1}{40 \times 10^3} = 25 \mu s$. This means that the 256 samples must represent 3 times this period. The sampling period is thus: $T_s = \frac{1}{f_s} = \frac{3 \times T_0}{L}$. The time axis can be generated using $t = (0 : L-1) * T_s$. So the following program can be written:

```
>> L = 256;
>> f0 = 40e3;
>> T0 = 1/f0;
>> Ts = (3*T0)/L;
>> fs = 1/Ts;
>> t = (0:L-1)*Ts;
>> x=sin(2*pi*f0*t);
>> plot(t,x);
```



The above figure is the result of the plot command. The horizontal axis is the vector *t* while the vertical axis is the vector $x = \sin(2\pi f_0 t)$.

More information can be obtained from the following web sites:

www.mathworks.com/products/matlab/

en.wikipedia.org/wiki/**MATLAB**

www.math.ufl.edu/help/matlab-tutorial/

The experiment

Part I

In this part, we are going to generate sinewaves and display them in the time and frequency domain.

1. Generate and plot 256 samples of a sinewave at 40 kHz. The sampling frequency is selected to be f_s = 3 MHz. How many periods of the signal do these 256 samples represent?

2. Use the FFT program (see MATLAB help) and represent the signal in the frequency domain between 0 and $f_s/2$. In order to generate the frequency axis, we can use the following MATLAB function: linspace. It generates linearly spaced vectors.

Syntax

```
y = linspace(a,b)
y = linspace(a,b,n)
```

Description

The `linspace` function generates linearly spaced vectors. It is similar to the colon operator ":", but gives direct control over the number of points.

`y = linspace(a,b)` generates a row vector `y` of 100 points linearly spaced between and including `a` and `b`.

`y = linspace(a,b,n)` generates a row vector `y` of `n` points linearly spaced between and including `a` and `b`. For `n < 2`, `linspace` returns `b` (taken from the MATLAB help). In our case, the correct statement should be:

>> f = (fs/2)*linspace(0,1,NFFT/2 + 1);

NFFT is the number of sample points and also the number of frequency samples. You can remark that this number is a power of 2. This is due to the fact that the FFT is computed more efficiently in this case.

Display the amplitude spectrum in *dBV* using the plot command.

The result of the FFT program is a row vector of complex numbers. So, if x is the signal in the time domain, then the following statements can be used:

>> Y = fft(x)/NFFT;

>>YdB = 20*log10(abs(Y(1:NFFT/2+1)));

You can remark that the vector YdB contains only NFFT/2+1 elements.

What is the frequency resolution? Is it possible to observe the 40 kHz sinewave in frequency domain?

3. In order to represent correctly the spectrum, we must use a much smaller sampling frequency. If we use the whole frequency range to represents frequencies between zero and four (4) times the sinewave frequency, we must have $\frac{f_s}{2} = 4 \times f_0 = 4 \times 40\,\text{kHz} = 160\,\text{kHz}$. So repeat 1 and 2 using f_s = 320 kHz.

How many periods will 256 samples represent? What is now the frequency resolution?

Write a small conclusion about the contradiction of having a good time resolution (1 and 2) and a good frequency resolution (3).

4. In this part, we are going to show the effect of undersampling, i.e. using a sampling frequency smaller than twice the carrier. Let us set f_s = 60 kHz and repeat 1 and 2.

What do you observe? What is the value of the frequency observed on the plot?

Part II

In this part, we are going to analyse the effect of the window on the resolution and also the leakage phenomenon.

1. Let us use a sampling frequency of 320 kHz but with a number of points NFFT equal to only 64. Generate 64 samples of the following signal:

$x(t) = \cos 2\pi f_0 t + \cos 2\pi f_1 t$ with f_0 = 40 kHz and f_1 = 43 kHz.

Use the rectangular window (i.e. no windowing) and represent the amplitude spectrum in *dBV*.

2. Use now a Hanning window. The following statement generates a column vector of N elements representing the different values of the Hanning window.

>> w = hann(N);

Since it is a column and not a row vector, it must be transposed before being applied to the signal x.

>> x_windowed = w' .*x;

Repeat 1.

3. Use a flat top window and repeat the above. The window is generated using the function flattopwin(N).

Is it possible to observe correctly the two frequencies?

4. With the same number of samples, generate the signal $x(t) = \cos 2\pi f_0 t + 0.1 \cos 2\pi f_1 t$, where f_0 = 40 kHz and f_1 = 32 kHz.

Use the FFT with the rectangular window, the Hanning window and the flat top window. Derive the amplitude of the components of the above signals.

Write a small conclusion about the two phenomena (resolution and leakage).

Write an overall conclusion.

You can develop the different MATLAB programs using the M file editor. At that time, you will be able to save your programs. Ask your lab assistants about the use of the editor.

Lab 2

Time and Frequency Measurements

Purpose

The objective of this experiment is to learn how to use the digital storage oscilloscope GDS1042 in order to display waveforms in the time and in the frequency domain.

The GDS1042 Oscilloscope

The GDS1042 is a digital storage oscilloscope. This means that the waveform is acquired, sampled, converted to numbers using an 8 bit analog to digital converter and finally stored into a buffer before it is displayed on an LCD screen. It is clear that its functions are quite different from a classical oscilloscope that uses a cathode ray tube (CRT). It is able to display signals that extend up to 40 MHz. Its input impedance is 1 MΩ if we use coaxial cables or a ×1 probe. This impedance increases to 10 MΩ if we use a ×10 probe. The time base can be set from 1ns/Div to 10s/Div. The sampling rate can go as high as 250 MHz (250 Mega Samples/s). The waveform can be stored in memories and the maximum size of a record is 4 k samples maximum (1 k = 1024).

The AFG-2125 arbitrary function generator

The AFG-2125 Arbitrary Function Generator is a DDS (Direct Digital Synthesized) based signal generator. It can accurately generate arbitrary waveform using a sampling rate of 20 MSamples/s, 10 bit resolution and 4k point memory. It can generate of Sine, Square (Pulse), Ramp (Triangle), Noise and Arbitrary waveforms. It includes a frequency counter for precision frequency measurements.

Instek AFG-2125 Features:

- 0.1Hz to 25 MHz with 1Hz Resolution
- Sine, Square, Triangular, Noise and Arbitrary Waveform
- 20MSa/s Sampling Rate, 10 bit Vertical Resolution and 4k Point Memory for Arbitrary Waveform
- 1% ~ 99% adjustable duty cycle for Square Waveform
- Waveform Parameter Setting Through Numeric Keypad Entry & Knob Selection
- Amplitude, DC Offset and Other Key Setting Information Shown on the 3.5” LCD Screen Simultaneously
- AM/FM/FSK Modulation, Sweep, and Frequency Counter Functions

- USB Device Interface for Remote Control and Waveform Editing
- PC Arbitrary Waveform Editing Software

<u>The Experiment</u>

Part I

1. Use the AFG-2125 to generate a sinewave at 40 kHz with a peak to peak amplitude of 10V. In order to do so, proceed as follows:

   1. Press the FUNC key to select the sine wave
   2. Press the FREQ key and enter 4, 0, kHz from the keyboard
   3. Press the AMPL key and enter 1, 0, Vpp from the keyboard
   4. Press the OUTPUT key in order to have an output from the generator.

Input the signal to the Channel 1 of the GDS1042. Use the AUTOSET button to measure the waveform in the time domain.

2. Press the MATH button of the oscilloscope to activate the FFT computation. The amplitude is displayed in red on the LCD screen. Select the Flat Top window. Can you distinguish the 40kHz sinewave in the above spectrum?

3. Press the ACQUIRE button and record the sampling frequency. Record the time base setting. Knowing that the sampling period and the time base setting are proportional, derive the relationship between the sampling frequency and the time base setting?

4. In order to be able to visualize correctly the spectrum, you must decrease the sampling frequency (Increase the time base setting). Increase the time base setting until the frequency display shows a peak in approximately the middle of the range (a setting of approximately 1ms/div will be ok).

* Measure the amplitude (in dBV) and the frequency of the waveform using the cursor measurements provided by the GDS1042. The reading should be $20\log_{10}\frac{A}{2}+\mathrm{k}$. The constant k depends on the chosen window. What is its value?

Compare with the time domain measurements.

* Change the window to Hanning. Compute the constant k for this window.

* Repeat the same measurement using the rectangular and the Blackman window.

Part II

1. Switch the waveform generated by the AFG-2125 to a square wave with an amplitude of 10V peak to peak. (The procedure is the same as for the sinewave but the duty cycle must be set at 50%).

    1. Press the FUNC key to select the square wave
    2. Press the FREQ key and enter 4, 0, kHz from the keyboard
    3. Press the AMPL key and enter 1, 0, Vpp from the keyboard
    4. Press the DUTY key and enter 5, 0, % from the keyboard
    5. Press the OUTPUT key in order to have an output from the generator.

2. Adjust the time base setting in order to display at least 3 harmonics. Using cursors, measure the frequency and amplitude of each component of the waveform.

3. Repeat your measurements using a triangular wave with the same peak to peak amplitude and 50% duty cycle.

4. Compare with the theoretical results given by the Fourier series development of both waveforms.

Write an overall conclusion.

Lab 3

Amplitude Modulation

Purpose

The objectives of this experiment is to measure important characteristics of Amplitude Modulated waves in the time and in the frequency domain

Discussion

The mathematical expression of an AM wave is:

$$x(t) = A[1 + ms(t)]\cos 2\pi f_0 t$$

where f_0 is the frequency of the carrier and $s(t)$ is the modulating signal. It is assumed that the modulating signal is bounded to a peak value of unity. Figure 1 below shows an AM with a sinusoidal envelope.

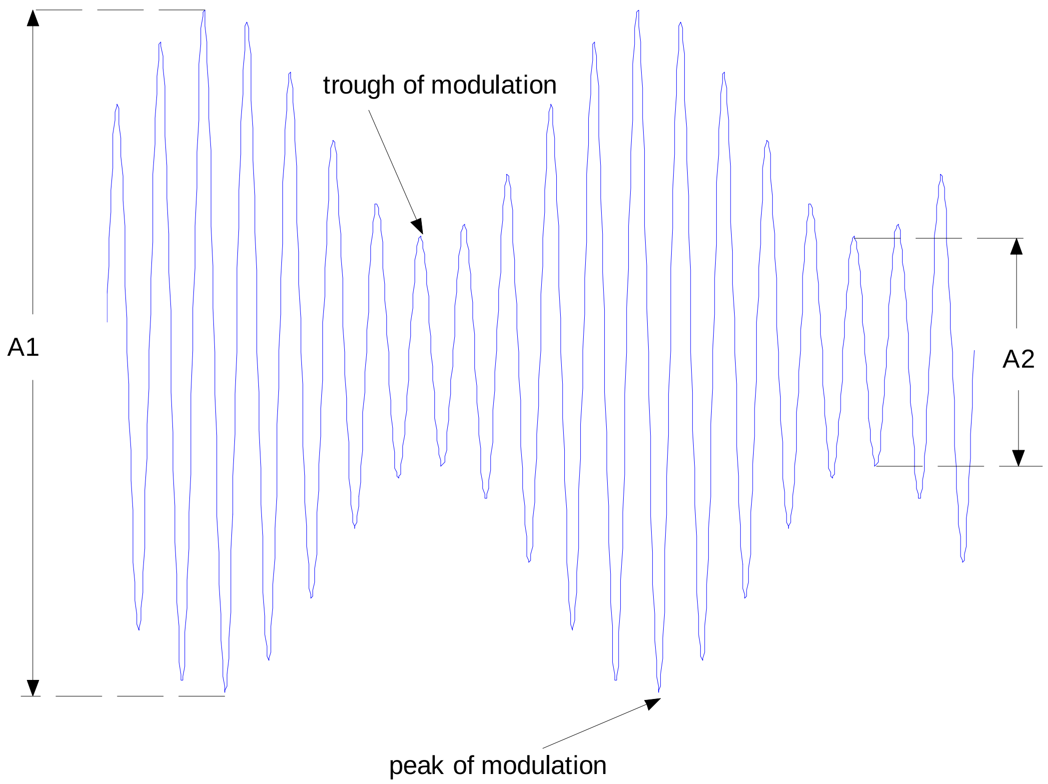


Figure 1 Typical AM wave

From the above figure, we can see that at a peak, we have A1 = 2A(1+m) and that at a trough, we have A2 = 2A(1-m), so we can deduce from the above that the carrier amplitude is:

$$A = \frac{A1 + A2}{2}$$

The modulation index is then:

$$m = \frac{A1 - A2}{A1 + A2}$$

The Experiment

Part I

1. For that part, we are going to generate a 40 kHz sinewave modulated by a 1 kHz low frequency sinewave from the AFG-2125. In order to do so, we must follow the subsequent procedure:

1. Press the FUNC key to select the sine wave
2. Press the FREQ key and enter 4, 0, kHz from the keyboard
3. Press the AMPL key and enter 1, 0, Vpp from the keyboard
4. Press the AM key
5. Press the Shift key and then the INT/EXT key and select the INT (internal) source
6. Press the Shift key and then the Shape key to select the sine wave
7. Press the Shift key and then the Rate key and enter 1, kHz from the keyboard
8. Press the Shift key and then the DEP/DEV key and enter 7, 0, % from the keyboard (this will produce a modulation index or depth of 70%)
9. Press the OUTPUT key in order to have an output from the generator.

Connect the BNC output of the generator to the channel 2 of the GDS1042 (this is the modulated signal) and connect the MOD output (top left BNC connector on the back of the AFG-2125) to the channel 1 of the GDS1042.

2. Set the time base of the oscilloscope (GDS1042) se that you can observe approximately 2 periods of the modulating signal (about 250 $\mu s/div$). Adjust the amplitude control of the oscilloscope so that you can observe on the screen of the GDS1042 a figure like the one you see in figure 1.

* Measure the modulation index.

3. Set now the GDS1042 to the XY mode. You are observing a Lissajou figure characteristic to AM modulation. The figure is a trapeze.

* Explain the figure.

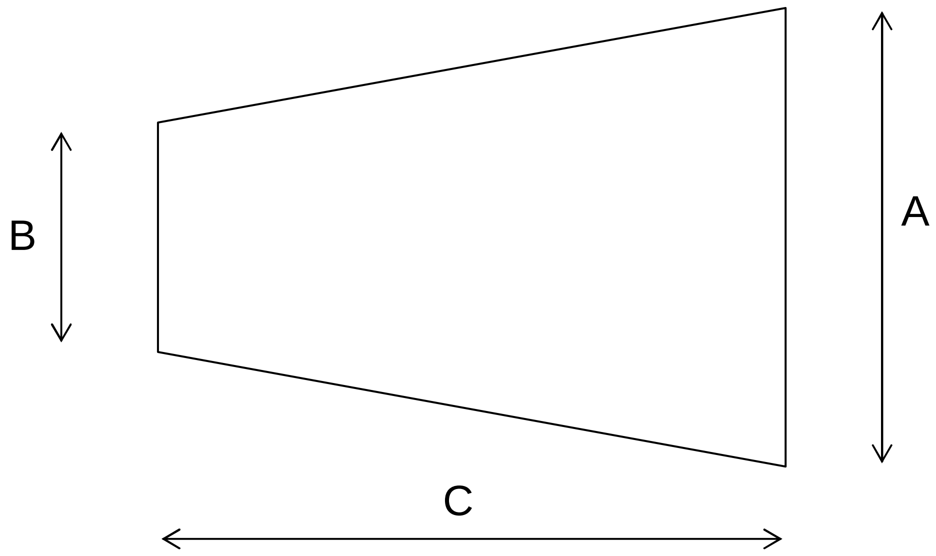


Figure 2

* Clearly explain in terms of the different waveforms ($\tilde{s}(t) = A_m \cos\omega_m t$ and $x(t) = A_0(1 + m\cos\omega_m t)\cos\omega_0 t$) the significance of the 3 variables A, B and C.

* Adjust the modulation depth of the AFG-2125 to 100%. The trapeze that you are observing is transformed into a triangle. Explain. Increase the value of the modulating signal to 120% (this is the maximum possible value), you will get overmodulation. In this case, what do you observe on the screen?

<u>Part II</u>

* Set now the low frequency at 5 kHz. Keep the carrier at 40 kHz , set back the modulation depth at 70% and set the time base at 1 ms/div. Repeat the XY measurements and compute the modulation index.

- Press the MATH button and set the input of the FFT (with the flat top window) to channel 2.
- Sketch the displayed spectrum.
- Are the frequency domain measurement matched to the time domain ones?
- Are your measurements compatible with theory?

Write an overall conclusion.

When you finish the experiment, press on the AM key to deselect the AM function.

Lab 4

Amplitude Demodulation

Purpose

The objective of this experiment is to implement correctly an AM demodulator using a peak detector.

Discussion

In this experiment, we are going to use a simulation software: Multisim® from National Instruments. This software is based on SPICE (Simulation Program with Integrated Circuit Emphasis). It extracts automatically the differential equations satisfied by the circuit under analysis and then solves them in the time domain. The equations are generated from the NETLIST (Network List) extracted from the schematic.

Part I

You will have to implement the circuit shown below.

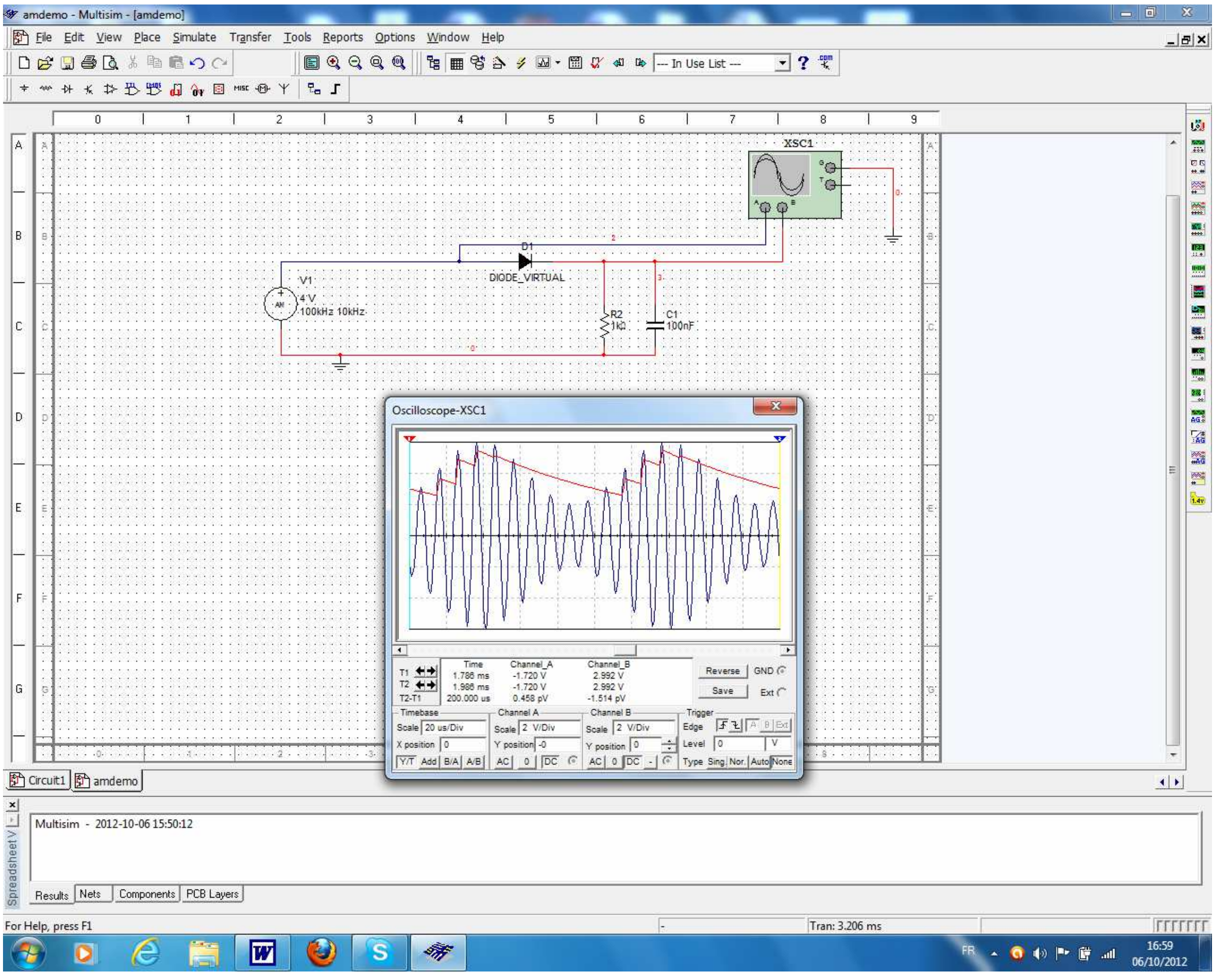


Figure 1 Multisim screen

In order to do so, you will have to follow the subsequent steps.

*** Saving the file:**

1. Select File/save to display a standard Windows Save dialog.
2. Navigate to the location where you want to save the file, and enter "amdemo" as the filename, and click on the Save button.

*** Placing the components:**

1. Select Place/component (ctrl-W) to display the Select a Component Browser, navigate to the group "sources" and select "signal_voltage" in the "family" group. Then, select "AM_Voltage" and press the OK button. The source will appear as a "ghost" on the cursor.
2. Move the cursor somewhere on the left of the screen. Press the left button of the mouse to deposit the source on the screen. Double click on the source to modify its parameters. Set its amplitude to 4 Volts, the carrier frequency at 100 kHz, the modulation index at 0.5 and the intelligence frequency (modulating frequency) at 10 kHz. Select the ground from the group "sources" and the family "power sources" and place it as shown in figure 1.
3. Place the diode (from the group "diodes" and the family "diode virtual").
4. Place the resistance from the group "basics" and the family "Basic_virtual". The resistance is horizontal. Once placed, press the right button of the mouse to rotate it. Double click (left) on it to set its value at 1 kΩ. Repeat the same operation to place the capacitance. Set its value at 100 nF.
5. On the right side of the screen, you can see a display of many instruments. Select the Oscilloscope from the list.

*** Wiring the Circuit:**

As soon as the cursor is over a pin, Multisim knows you want to wire and the pointer changes to a crosshair.

1. Click on a pin on a component to start the connection and move the mouse. A wire appears, attached to the cursor.
2. Click on the destination pin to finish the connection.
3. Connect completely the circuit as shown in figure 1.

*** Simulation:**

Double click on the oscilloscope icon. Set the control of the oscilloscope as shown in figure 1. Press the "start simulate" button () to start the simulation of the circuit. Observe the waveform displayed on the oscilloscope. Stop the simulation by pressing the same button.

* Sketch the displayed waveform. Compare the time constant RC with the period of the modulating waveform and the carrier waveform. In order to change the color of the displayed waveform (on channel A), right click on the wire leading to the channel A input and change its color. This will help discriminating between waveforms. Is there any distortion visible on the demodulated waveform?

* Change the value of the capacitance to 10 nF. Sketch the displayed waveform. Compare the time constant RC with the period of the modulating waveform and the carrier waveform.

* Change the carrier frequency to 1000 kHz. Sketch the displayed waveform. Compare the time constant RC with the period of the modulating waveform and the carrier waveform.

Write an overall conclusion.

Lab 5
FM Spectra

Purpose

The objective of this experiment is to verify the mathematical analysis of a frequency modulated wave. First you will determine the characteristics of the modulator. Then you will apply various types of modulating signals to the system and observe the spectra which are present.

Discussion

In this experiment, the students are going to generate Frequency Modulated sinewaves. The AFG-2125 is capable of producing frequency modulated waveforms. We have full choice of carrier and modulating waveform. The generator is fully programmable. We have complete control on the carrier shape, frequency and amplitude. At the same time, we can adjust the shape of the modulating signal, its frequency and also the maximum frequency deviation Δf. The instantaneous frequency is $f(t) = f_0 + (\Delta f)s(t)$ and if the modulating waveform is a sinewave of frequency $f_m = \dfrac{\omega_m}{2\pi}$,the FM waveform is: $x(t) = A\cos\left(\omega_0 t + \beta \sin \omega_m t\right)$ with $\beta = \dfrac{\Delta f}{f_m}$.

The experiment

Part I

In this part of the experiment, we are going to set the modulating frequency at a very low value so that we can observe the variations of the frequency directly on the oscilloscope. We are going to generate a sinusoidal carrier at 100 kHz modulated by a low frequency signal having a carrier of 1 Hz. This low value will give you enough time to observe the modulation on the screen.

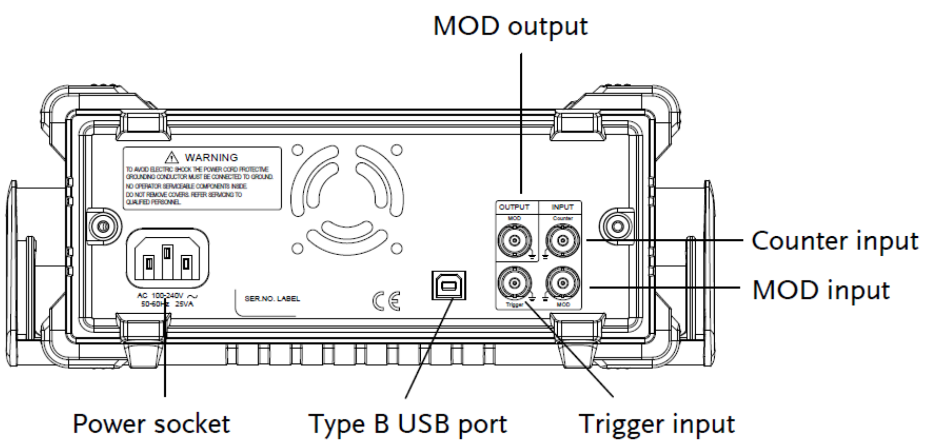
1. Press the FUNC key to select the Sine wave.
2. Set the carrier frequency at 100 kHz using the FREQ key and the keyboard.
3. Set the amplitude at 1 V_{pp} using the AMPL key. (Make sure that the output impedance is set at 50 Ω)
4. Press the FM key.
5. Select the internal source by pressing Shift, then INT/EXT and select INT source.
6. Select a sine modulating by pressing Shift, then Shape repeatedly to select Sine wave.

7. Set its frequency at 1 Hz by pressing Shift then rate then 1 on keyboard then Hz/Vpp key.
8. set the deviation at 10 kHz by pressing Shift then DEP/DEV key then 1,0 on the keyboard then kHz/Vrms key.
9. Press the OUTPUT key to observe the waveform on the oscilloscope.
10. Set the time base on the oscilloscope to 10 µs/div and observe the output of the AFG-2125 on the oscilloscope. You are seeing an FM wave.
11. Change the shape of the modulating waveform to triangular and then to square. Try to explain in your own mind the reasons for the different wave shapes.

Part II

1. Set the AFG-2125 generator at 100 kHz, sine wave, 1 V peak to peak output, with no modulation. Connect it to the channel 1 input of the oscilloscope. Set the time base at 250 µS/Div. This will set the sampling rate at 1 MHz. Using the MATH operation, set the GDS1042 to FFT display with the flat top window. Set the vertical scale of the FFT to 10 *dB*/Div. Record the carrier level without modulation using the Y cursor.
2. Set now the AFG-2125 to FM modulation. The modulating signal must be sinusoidal at 7 kHz. Set the frequency deviation Δf at 21 kHz. Use a BNC to BNC cable to connect the modulating signal (from the MOD output on the rear panel) to channel 2 of the GDS 1042 to observe it. See the figure below.

AFG-2105/2112/2125 Rear Panel



3. Record the amplitude of the carrier of the modulated waveform. The ratio of the amplitudes of the carrier with and without modulation should give you the value of $J_0(\beta)$. From a table of Bessel functions, determine the value of β. Deduce the value of Δf. It is the same value that you have set?
4. Adjust the value of Δf in order to have $\beta = 2.40$. What do you observe?
5. Repeat question 4 but set Δf in order to have $\beta = 3.83$.

Write an overall conclusion.

Note: $J_0(\beta) = 0$ for $\beta = 2.40, 5.52$. $J_1(\beta) = 0$ for $\beta = 3.83, 7.02$

Lab 6

FM Demodulation

Purpose