

## ANALOGIC Protoboard....

The Programmable Analog Solution

### List of Experiments

Here is the list of experiments which can be performed using the ANALOGIC Protoboard. These assignments can be used to teach beginners that “How to use FPAA Technology”.

On the next pages we have also given one assignment from the list.

1. LPF Filter
2. HPF Filter
3. AM Modulation
4. AM Demodulation
5. ASK
6. FSK
7. PSK
8. Comparator
9. Rectifier
10. Integrator
11. Differentiator
12. Peak Detector
13. PID Controller
14. VCO
15. PWM
16. Zero Crossing Comparator
17. Bilinear Filters
18. Phase Shifters

... and many more

## Amplitude Modulation

### Aim:-

To design Amplitude Modulator (AM) using FPPA.

### Theory:-

In amplitude modulation, the amplitude of the carrier varies in accordance with the instantaneous value of the modulating voltage.

Let the carrier signal is

$$V_c = V_c \cos \omega_c t$$

and the modulating signal is

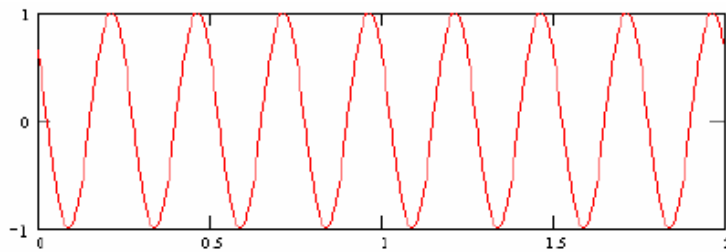
$$V_m = V_m \cos \omega_m t$$

Hence the modulated signal is

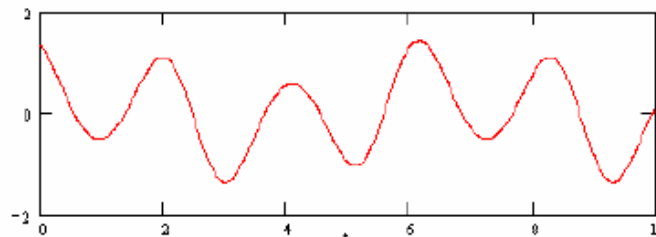
$$\begin{aligned} V_c' &= (V_c + k_a V_m \cos \omega_m t) \cos \omega_c t \\ &= V_c \cos \omega_c t + k_a V_m \cos \omega_m t \cos \omega_c t \\ &= V_c (1 + m_a \cos \omega_m t) \cos \omega_c t \end{aligned}$$

where

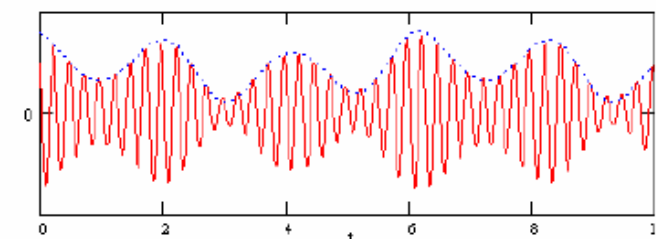
$m_a = k_a V_m / V_c$  is known as the depth of modulation or modulation index.. When  $m_a$  is expressed in percentage, it is called percentage of modulation. The amplitude of modulated carrier is given by  $V_c (1 + m_a \cos \omega_m t)$ .



Carrier Signal



Modulating Signal

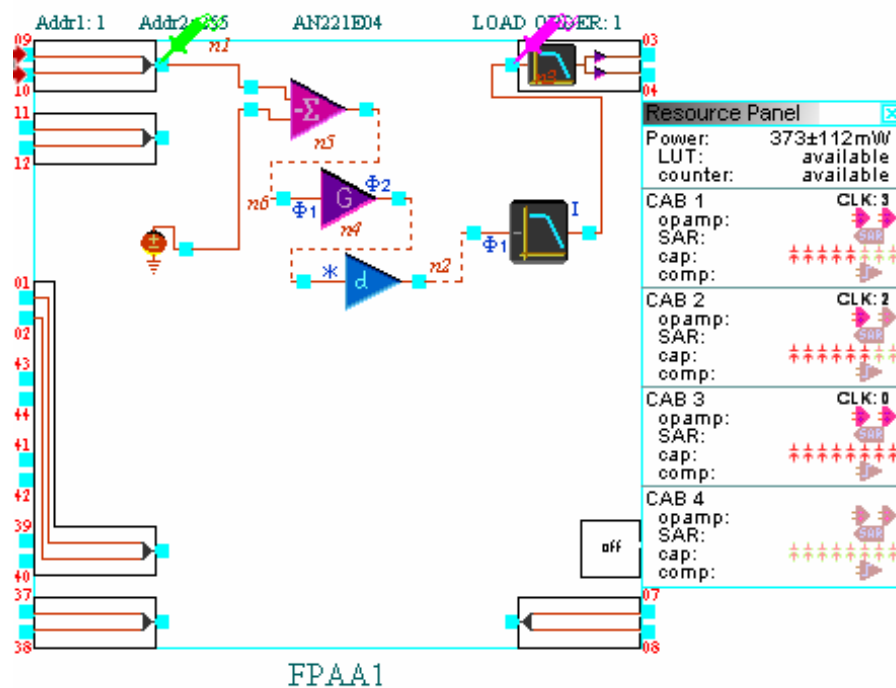


Amplitude modulated signal

The value of modulation index of amplitude modulated signal should not be greater than 1. If the modulation index is greater than 1, then it is called over modulation index. If modulation index is less than 1, then it is called under modulation. It is difficult to recover the modulating signal from an over modulated signal, so all the amplitude modulated signal should be under modulated.

**Implementation in FPAA:-**

Field Programmable analog arrays are sampled data system. So all the CAM's (Configurable analog modules) used will have a sampling clock. The master clock frequency for the design is 16MHz. The CAM's that are used for the AM design are Inverting Sum Stage, Half Cycle Gain and Inverting Differentiator. The design of AM modulator is shown below.



In this design AM modulation is actually carried out by Half Cycle Gain Cam and inverting differentiator CAM. The sampling clock of Half Cycle gain is set to 100KHz. This Cam is deciding the carrier frequency. In this example the carrier frequency is set to 100KHz. This CAM samples the incoming modulating on the phase 1 of carrier frequency. The output of this CAM reflect the modulating signal only during the phase2 of the sampling clock and during the 1<sup>st</sup> phase of the sampling clock the output of the CAM will be at the VMR reference voltage. Thus the output from this Cam will be a chopped base-band signal

This chopped base-band signal is fed into an inverting differentiator. In a continuous time system, the output of a differentiator circuit at a particular instant will be proportional to the slope of the input signal at that same instant. But in a sampled system like FPAA, the output signal will be proportional to the change in voltage ( $\Delta V$ ) over the change in time ( $\Delta T$ ). The "slope" during this sample is  $\Delta V/\Delta T$ . The differentiation constant k of the inverting differentiator sets the gain for the circuit. The output of the inverting differentiator CAM is given by

$$V_{out} = -k \Delta V/\Delta T$$

In this case the differentiation constant 'k' is set to 1  $\mu$ sec. The low value of k ensures that the output of the rail will not be clipped for large amplitude input signals. During the 1<sup>st</sup> phase, the input signal slope will be positive which produces a negative step at the differentiator. In the very next phase, the slope will have the same magnitude, but opposite sign, and so the differentiator output will now be a positive step of the same amplitude. The result of all this will be amplitude modulated signal centered about VMR, with each phase's amplitude proportional to the input signal. In plainer language, we now have an AM signal.

A summing amplifier is used to provide a DC offset to the input signal to keep it always above VMR. Failing to do so will result in an over-modulated signal (modulation index > 1) which precludes the use of simple envelope detection for demodulation. A much more complex synchronous demodulation scheme would otherwise be required. This is rarely used in AM receiver circuits.

At the output stage a low pass filter with a corner frequency in the range of sampling frequency, so as to filter out high frequency components from the output signal. The designer can also include a smoothing filter at the output stage having a corner frequency of 100 KHz.

The design can be verified using the simulator provided by Anadigm Designer® 2 software. Once satisfied, the designer can download the design to FPAA.

### Parameters

#### Inverting Sum Stage

Sampling Clock = 100 KHz  
Lower input Gain = 1  
Upper input Gain = 0.666

#### Half Cycle Gain Stage

Sampling Clock = 100 KHz  
Input Sampling Phase = Phase 1  
Gain = 1.5

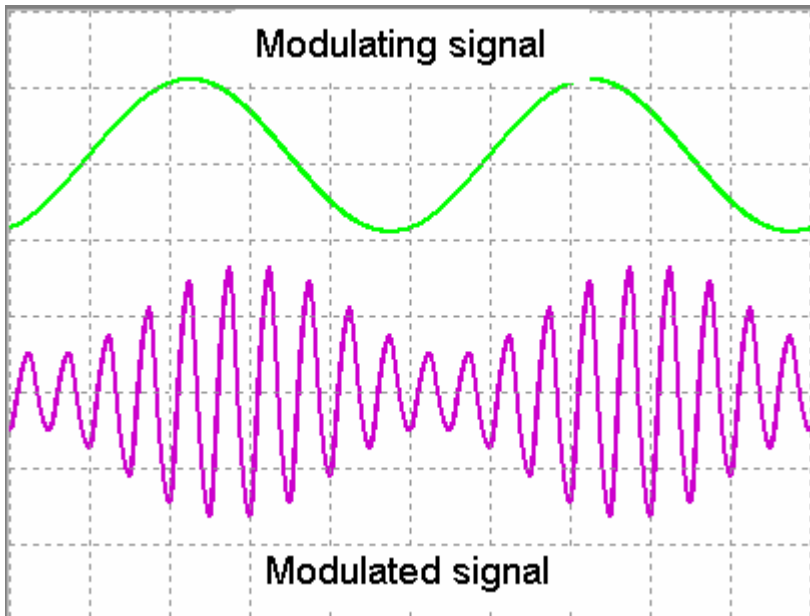
#### Inverting Differentiator

Sampling Clock = 100 KHz  
Output Hold = Off  
Differentiation Constant = 1  $\mu$ sec

#### BiQuadratic Filter Stage

Sampling Clock = 2 MHz  
Corner Frequency = 110KHz  
Gain = 1  
Quality Factor = 0.707

## Results



The AM modulator has been downloaded into FPAA and its outputs are compared with the simulator result.

Note: You can use following sample code for above experiment.

AM modulator.ad2