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Direct Digital Synthesizer

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Abstract - *Direct Digital Synthesizer (DDS) is the more advanced function generator which can solve the problems of conventional function generators to a great extent. It has an Increasing role in digital clock and agile clock generation, and modulation. DDS roots from sampling and quantizing theorems and generate wide range of frequencies with distortion less than -55dB below 50 kHz and less than -40dB above 50 kHz. Also DDS can support Amplitude modulation, frequency modulation, or phase modulation. DDS uses a fixed-frequency clock and a simpler filtering scheme, so it's less expensive than the Points per Clock (PPC) method. In DDS, a phase accumulator a DDS an increment to its output in every clock cycle, and the accumulator's output represents the phase of the waveform. The output frequency is proportional to the increment, so it's easy to change frequency even though the clock frequency is fixed. The output of the accumulator is converted from phase data into amplitude data typically by passing it through some type of look -up table. The phase accumulator design allows DDS to use a fixed clock, but still execute waveforms at a perceived faster sample rate than the clock. With DDS, not every individual point in the waveform memory is being expressed in the resulting output waveform. Instead, DDS outputs a best approximation of the waveform, which means small features in the waveform can be partially or completely skipped over.*

***Key Words*: Direct Digital Synthesis, Points Per Clock, Arbitrary Waveform Generators, Numerically Controlled Oscillator, Vector Signal Generators, Spurious Free Dynamic Range, Thin Shrink Small Outline Package.**

1. INTRODUCTION

A function generator is usually a piece of electronic test equipment or software used to generate different types of electrical waveforms over a wide range of frequencies. Some of the most common waveforms produced by the function generator are the sine wave, square wave, triangular wave and saw tooth shapes. These waveforms can be either repetitive or single-shot (which requires an internal or external trigger source). Integrated circuits used to generate waveforms may also be described as function generator ICs.

In addition to producing sine waves, function generators may typically produce other repetitive waveforms including saw tooth and triangular waveforms, square waves, and pulses. Another feature included on many function generators is the ability to add a DC offset. Although function generators cover both audio and RF frequencies, they are usually not suitable for applications that need low distortion or stable frequency signals. When those traits are required, other signal generators would be more appropriate. Some function generators can be phase-locked to an external signal source (which may be a frequency reference) or another function generator .Function generators are used in the development, test and repair of electronic equipment. For example, they may be used as a signal source to test amplifiers or to introduce an error signal into a control loop. Function generators are primarily used for working with analog circuits, related pulse generators are primarily used for working with digital circuits.

Conceptually, the simplest way to generate a waveform is to store its points in memory and then read those points out one after another and clock them into a DAC. After the last point has been read, the generator jumps back to the first point again to begin the next cycle. This is sometimes called "point per clock" (PPC) generation.

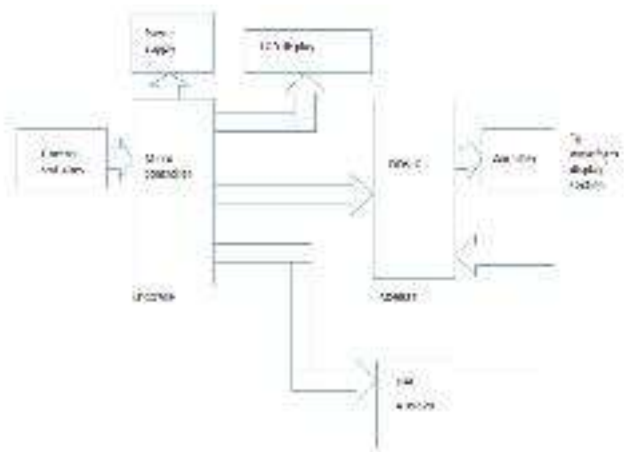
Direct digital synthesis (DDS) is a method employed by frequency synthesizers used for creating arbitrary waveforms from a single, fixed- frequency reference clock. DDS is used in applications such as signal generation, local oscillators in ommunication systems, function generators, mixers, modulators, sound synthesizers and as part of a digital phase-locked loop.

This frequency synthesizer technique is becoming more widespread. DDS, direct digital synthesis takes a different approach to that of the more usual indirect frequency synthesis techniques using PLLs by directly synthesizing the waveform from a digital map of the waveform stored in a memory. Using digital techniques in this way, along with high speed logic, direct digital synthesis provides a powerful technique for creating accurate signals whose frequency can be stepped by very small increments giving virtually analogue or continuous tuning if needed. For many years, direct digital synthesizers were limited in frequency by the speed of the logic. With speeds improving he top frequency limits for direct digital synthesizers is increasing.

Most new signal generators, such as arbitrary waveform generators (AWGs), lower-frequency RF generators, and vector signal generators (VSGs), use direct digital synthesis (DDS) instead of fractional-N phase-locked-loop (PLL) synthesizers, which are common in older instruments and higher-frequency RF generators.

A DDS uses the waveform-generation method with a sophisticated phase accumulator circuit. A clock signal from crystal-based oscillator clocks the adder and output address register/accumulator that feeds the address to the DAC. The address register output is added to a phase register value to get the next ROM address value. By changing the phase register value, which comes from a processor or other circuit, the phase increment of the output is changed. To alter the frequency, you change the phase register value.

2. GENERAL BLOCK DIAGRAM



Block diagram of a DDS function generator is shown above. The main sections of the design are power supply section, microcontroller section, DDS IC section, DAC section for amplitude control, control switches section, LCD display section which displays the mode, frequency and amplitude of the waveforms, amplifier section and finally a display section where we can see the waveforms.

The microcontroller for controlling the DDS digital ic is the ARM controller LPC 1769.ARM is basically a controller of RISC architecture. The LPC 1769 is an ARM Cortex-M3 based microcontroller for embedded applications featuring a high level of integration and low power consumption. The ARM Cortex-M3 is a next generation core that offers system enhancements such as enhanced debug features and a higher level of support block integration.

The DDS IC that we used in our project is AD9834. The AD9834 is a 75 MHz low power DDS device capable of producing high performance sine and triangular outputs. It also has an on-board comparator that allows a square wave to be produced for clock generation. Consuming only 20 mW of power at 3 V makes the AD9834 an ideal candidate for power-sensitive applications .The serial peripheral interface (SPI) is used to interface the LPC 1769 and AD9834.A DAC AD 5620 is also interfaced to LPC1769 through the SPI interfacing protocol for amplitude controlling of the waveforms. The AD5620/AD5640/AD5660, members of the *nano* DAC™ family of devices, are low power, single, 12-/14-/16-bit, buffered voltage-out DACs and are guaranteed monotonic by design. The four control switches we used are the push on push off buttons for controlling the different parameters of the waveform. One switch is for controlling different modes (sine, triangle and square), one for increasing the frequency, one for decreasing the frequency and one for increasing the amplitude.

A 16x2 LCD display (JHD162A) is used to display the waveform type, frequency and amplitude. We are using opamp amplifier circuits with high gain to amplify the output signals from DDS IC. Finally the amplified wave forms can be viewed using a Cathode Ray Oscilloscope (CRO) or Digital Storage Oscilloscope (DSO).

We use SPI (Serial Peripheral Interface) communication between ARM controller and AD9834 IC.

3. BASIC WORKING OF DDS

As the name suggests this form of synthesis generates the waveform directly using digital techniques. This is different to the way in which the more familiar indirect synthesizers that use a phase locked loop as the basis of their operation. A direct digital synthesizer operates by storing the points of a waveform in digital format, and then recalling them to generate the waveform. The rate at which the synthesizer completes one waveform then governs the frequency. The overall block diagram is shown below.

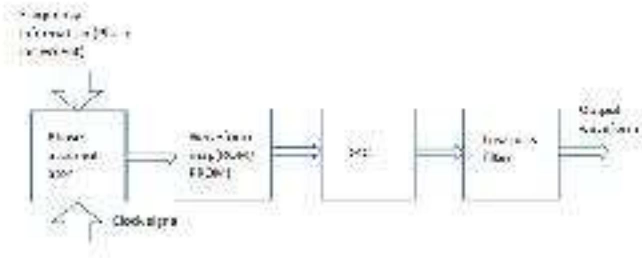


Figure 3.1: Basic direct digital synthesizer block diagram

The operation can be envisaged more easily by looking at the way that phase progresses over the course of one cycle of the waveform. This can be envisaged as the phase progressing around a circle. As the phase advances around the circle, this corresponds to advances in the waveform. The synthesizer operates by storing various points in the waveform in digital form and then recalling them to generate the waveform. Its operation can be explained in more detail by considering the phase advances around a circle as shown in Figure below.

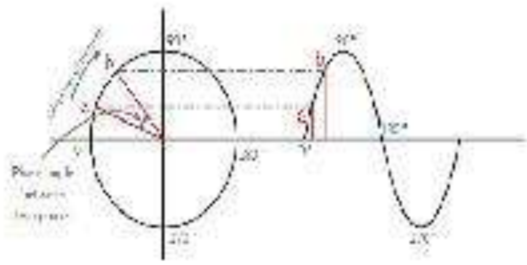


Figure 3.2: Phase angle of points on a sine wave

As the phase advances around the circle this corresponds to advances in the waveform, i.e. the greater the number corresponding to the phase, the greater the point is along the waveform. By successively advancing the number corresponding to the phase it is possible to move further along the waveform cycle. The digital number representing the phase is held in the phase accumulator. The number held here corresponds to the phase and is increased at regular intervals. In this way it can be sent hat the phase accumulator is basically a form of counter. When it is clocked it a DDS a preset number to the one already held. When it fills up, it resets and starts counting from zero again. In other words this corresponds to reaching one complete circle on the phase diagram and restarting again.

Once the phase has been determined it is necessary to convert this into a digital representation of the waveform. This is accomplished using a waveform map. This is a memory which stores a number corresponding to the voltage required for each value of phase on the waveform. In the case of a synthesizer of this nature it is a sine look up table as a sine wave is required. In most cases the memory is either a read only memory (ROM) or programmable read only memory (PROM). This contains a vast number of points on the waveform, very many more than are accessed each Cycle. A very large number of points is required so that the phase accumulator can increment by a certain number of points to set the required frequency.

The next stage in the process is to convert the digital numbers coming from the sine look up table into an analogue voltage. This is achieved using a digital to analogue converter (DAC). This signal is filtered to remove any unwanted signals and amplified to give the required level as necessary. Tuning is accomplished by increasing or decreasing the size of the step or phase increment between different sample points. A larger increment at each update to the phase accumulator will mean that the phase reaches the full cycle value faster and the frequency is correspondingly high. Smaller increments to the phase accumulator value means that it takes longer to increase the full cycle value and a correspondingly low value of frequency. In this way it is possible to control the frequency. It can also be seen that frequency changes can be made instantly by simply changing the increment value. There is no need to a settling time as in the case of phase locked loop based synthesizer.

From this it can be seen that there is a finite difference between one frequency and the next, and that the minimum frequency difference or frequency resolution is determined by the total number of points available in the phase accumulator. A 24 bit phase accumulator provides just over 16 million points and gives a frequency resolution of about 0.25 Hz when used with a 5 MHz clock. This is more than adequate for most purposes. These synthesizers do have some disadvantages. There are a number of spurious signals which are generated by a direct digital synthesizer. The most important of these is one called an alias signal. Here images of the signal are generated on either side of the clock frequency and its multiples. For example if the required signal had a frequency of 3 MHz and the clock was at 10 MHz then alias signals would appear at 7 MHz and 13 MHz as well as 17 MHz and 23 MHz etc.. These can be removed by the use of a low pass filter. Also some low level spurious signals are produced close in to the required signal. These are normally acceptable in level, although for some applications they can cause problems.

4. IMPLEMENTATION

4.1. SOFTWARE IMPLEMENTATION

We mainly used two software applications in our project.

1. Altium designer (version17.1.5)

This software application is used to create the schematics and the PCB layout of our circuit design.

2. LPCXPRESSO

This software application is used for programming LPC1769.

4.1.1. ALTIUM DESIGNER (VERSION 17.1)

Altium Designer is a PCB and electronic design automation software package for printed circuit boards. It is developed by Australian software company Altium Limited. Altium Designer version 6.8 from 2007 was the first to offer 3D visualization and clearance checking of PCBs directly within the PCB editor. Altium Designer's suite encompasses four main functional areas: schematic capture, 3D PCB design, Field Programmable Gate Arrays (FPGA) development and release/data management. Noteworthy features referred to in the reviews include:

- Integration with several component distributors allows search for components and access to manufacturer's data.

- Interactive 3D editing of the board and MCAD export to STEP.
- Cloud publishing of design and manufacturing data.

- Simulation and debugging of the FPGA can be achieved using the VHDL language and checking that for a given a set of input signals the expected output signals would be generated. FPGA soft processor software development tools (compiler, debugger, profiler) are available for selected embedded processors within an FPGA. Altium Designer is generally found to be more costly than other PCB design software but is noted for its ability to achieve fast results for complex circuits.

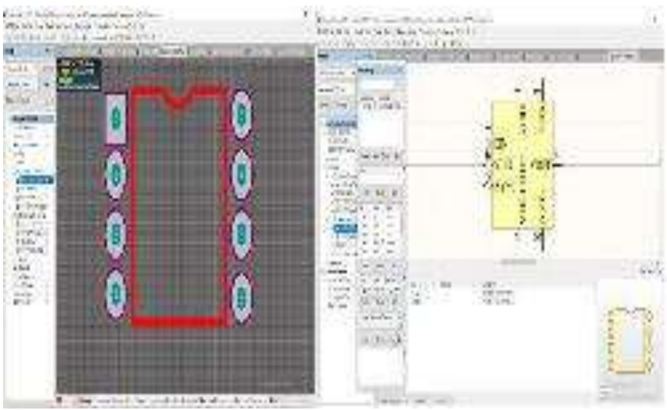


Figure 4.1: Symbol and footprint created for AD9834

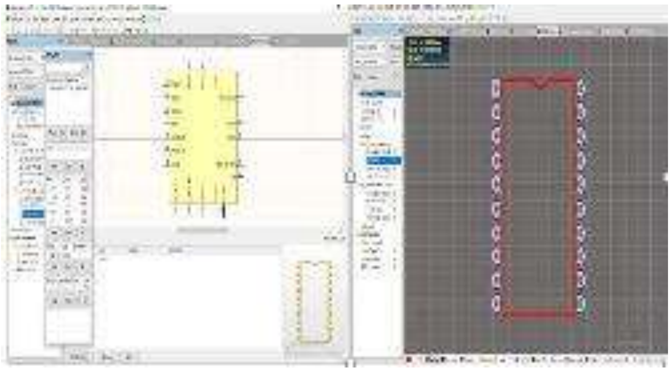


Figure 4.2: Footprint and symbol created for AD5620(DAC)

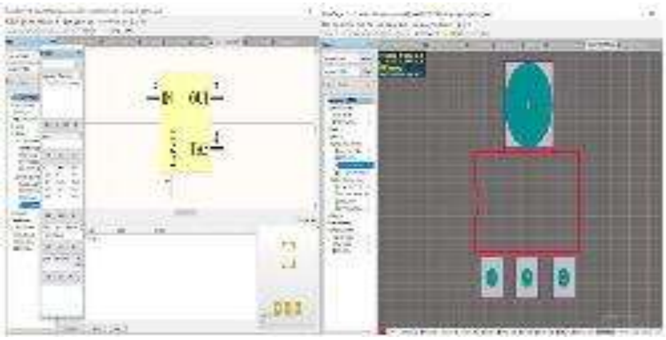


Figure 4.3: Symbol and footprint created for voltage regulator LD1117S333CTR

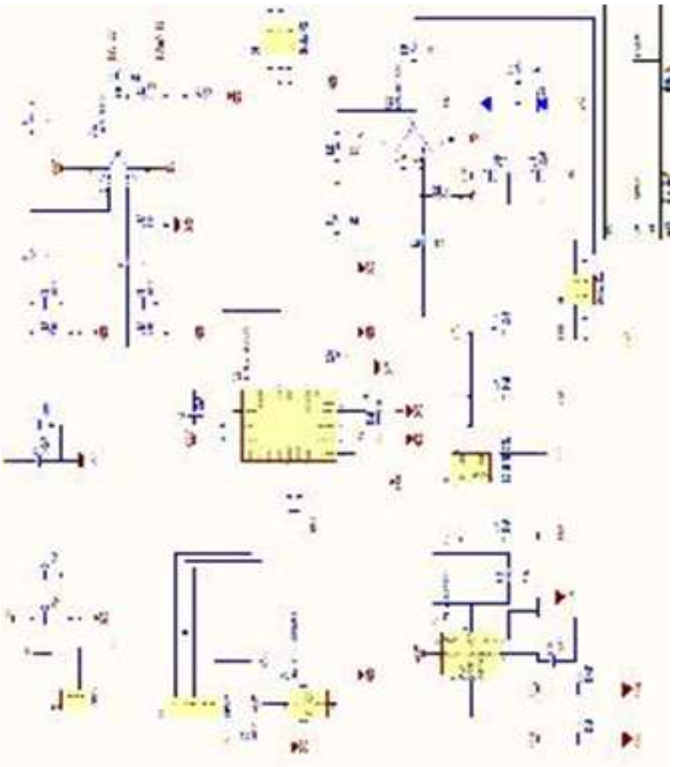


Figure 4.4: Schematic of DDS Board

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4.1.2. LPCXPRESSO

PCXpresso is a low-cost microcontroller (MCU) development platform ecosystem from NXP, which provides an end-to-end solution enabling engineers to develop embedded applications from initial evaluation to final production. The LPCXpresso platform ecosystem includes:

- The LPCXpresso IDE, a software development environment for creating applications for NXP's ARM based "LPC" range of MCUs.

- The range of LPCXpresso development boards, each of which includes a built-in "LPC-Link", "LPCLink2", or CMSIS-DAP debug probe. These boards are developed in collaboration with Embedded Artists.

- The standalone "LPC-Link2" debug probe. This guide is intended as an introduction to using LPCXpresso, with particular emphasis on the LPCXpresso IDE. It assumes that you have some knowledge of MCUs and software development for embedded systems.

We use LPCXpresso software application for the complete coding for LPC 1769 microcontroller.

4.2. HARDWARE IMPLEMENTATION

We have two separate PCB boards. First we implemented the DDS board corresponding to the above schematic(Figure 4.4).Then we implemented the LPC 1769 board which consists of the controller section ,control switches for varying the amplitude, frequency and mode of the waveforms and the interfacing headers to DDS board ,LCD display etc. We have the interfacing wires in between the LPC board and DDS board as well as the interfacing wires with LCD display. The hardware looks like the figure below;

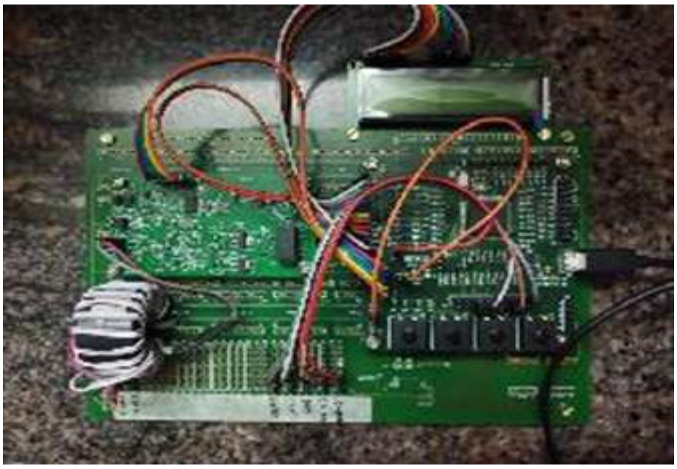

Figure 4.5: Completed hardware

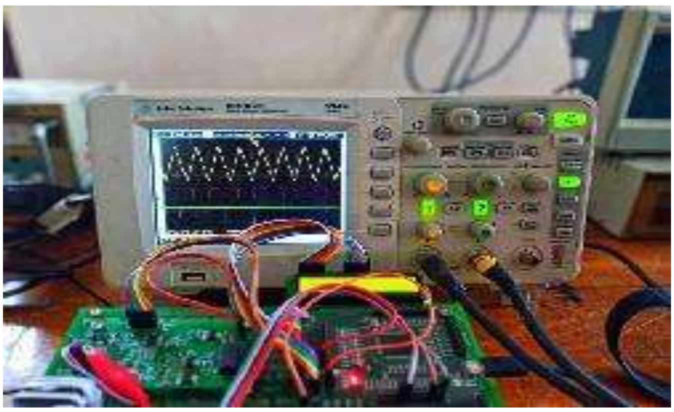

Figure 4.6: Triangular wave of 7 KHz

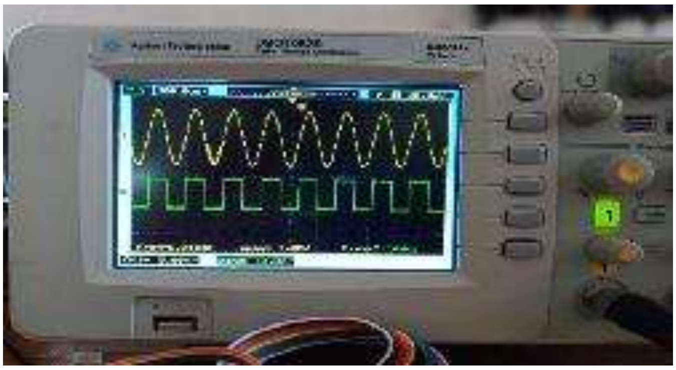

Figure 4.7: Sine and square wave of 7 KHz

5. CONCLUSIONS

DDS provides a way to generate analog signals from values stored in memory using digital techniques. By changing a tuning register. Frequencies can be changed quickly without setting time, making it ideal for testing, communication, waveform generation and frequency sweep applications. The project helps us to find out the different benefits of DDS function generators over the conventional function generators. The benefits of DDS are:

* The ability to generate arbitrary frequencies with accuracy and stability, limited only by the oscillator used to clock the phase accumulator.

* The frequencies provided by DDS are repeatable.

* High frequency resolution can be achieved with the digital techniques used in DDS.

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