Hybrid FM Stereo Encoder using DDS

for carrier and pilot signal generation

Manolis G.Tampouratzis [1]
[1] Department of Electronic Engineering
Faculty of Applied Sciences
Technological Educational Institute of Crete
Romanou 3 Chalepa, 73133 Chania, Crete, Greece
{tampouratzis@chania.teicrete.gr}

George A. Adamidis [2] (MSc in Electronic Physics)
[2] 1 E.K Heraklion – Technical School Laboratories
Division of Electronics
Itanou 40 Kipoupoli, 71307 Heraklion, Crete, Greece
{sv7fid@yahoo.gr}

***Abstract* - An fm-stereo generator device uses a complex modulation system, according to F.C.C standards, to achieve a compatible mono/stereo system of broadcasting. There are several approaches for building an FM-Stereo generator. In the current implementation, we present an hybrid FM-stereo generator which uses both digital and analog techniques. We use Direct Digital Synthesis (DDS) module for carrier and pilot tone generation which gives unlimited control over phase shift and the ability to produce clean (purely sinusoids) signals with great frequency accuracy and stability. Reference clock frequency (or crystal choice) is not very critical in a high resolution DDS and signal generation becomes simple, robust and completely accurate. Finally using DDS also diminishes the necessity of using complex (high order) filtering.**

Keywords *– direct digital synthesis (DDS) , fm stereo generator, pilot signal, carrier, balanced modulator, fm stereo spectrum*

I. INTRODUCTION

FM stereo broadcasting was introduced during the early 1960s. The fm stereo system which approved for use by the F.C.C in the U.S and later was adopted worldwide uses a complex modulation system to achieve a compatible mono/stereo system of broadcasting. Essentially, the system performs the multiplexing of two audio signals and further combines them into a complex baseband signal that modulates the FM carrier.

The system works by broadcasting a sum of the left (L) and right (R) audio channels, a pilot tone of 19 kHz and a double sideband suppressed carrier (DSBSC) sub-channel that contains the difference of the two audio channels (see fig. 1).

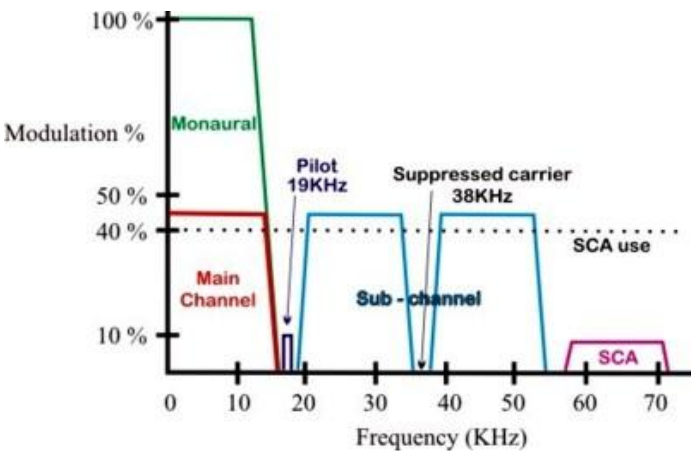


Fig. 1. The Composite FM-Stereo Spectrum

In a simple monaural system, the FM channel is frequency modulated ±75KHz with the audio information and the monaural audio signal occupies the 0-15KHz spectrum of the transmitted frequency spectrum (see figure 1). When stereo is transmitted, the same monaural signal (left plus right channel combined) remains in the 0-15KHz spectrum of the FM stereo signal and an additional sub – channel, centered at 38 KHz, which is a double sideband suppressed carrier signal (DSBSC) is additionally transmitted (see figure 1). This subcarrier is a left-subtracted-from-right (L-R) signal, which, when fed threw a matrix with the monaural main channel on the receiver, forms the individual left and right channels. An additional pilot career signal at 19 KHz is also transmitted. The pilot signal is phase-cohered (synchronized), to the suppressed 38 KHz carrier.

In an FM-stereo system, the monaural signal is modulated about 45%, the sub channel and the pilot tone are modulated 45% and 10%, respectively, so that the total modulation for a stereo FM- station is 100%. In modern stations where some SCA or RDS/RBDS subcarriers are also used, the modulation of the main and the sub channel are furthermore reduced in order to the total modulation being kept less than 100% (±75KHz deviation).

In an FM-stereo receiver the 19 KHz pilot signal indicates that the transmission is stereo. The receiver regenerates the 38 KHz carrier and then uses coherent detection for the sub-channel. Cohered detection only works when the carrier is present at the receiver. Off course, the receiver can not obtain the 38 KHz carrier from the baseband signal directly (because the carrier is suppressed during transmission). The carrier is actually obtained in the receiver from the 19 KHz pilot signal.

The composite FM-stereo signal that modulates the FM carrier in any FM-station is generated from a device which is often called as an "FM-Stereo Generator" or as an "FM-Stereo encoder". The typical theoretical diagram of an FM-stereo generator is shown on fig. 2

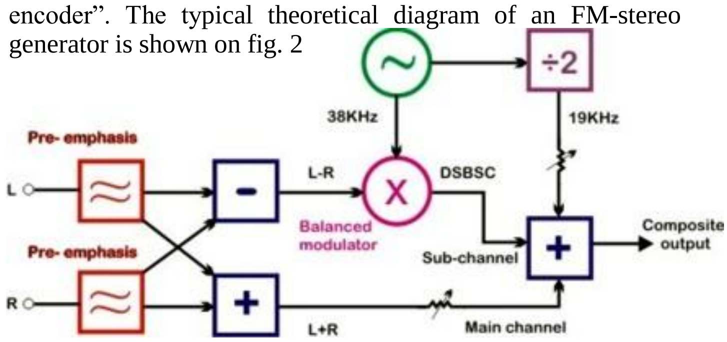


Fig. 2. Typical Theoretical diagram of an FM-Stereo Generator

ISBN: 978-1-61804-242-2

With respect to figure 2, both the left and the right audio channels are pre-emphasized, just as normal monaural signal would be. Then, the left and the right signals are both added and subtracted on a matrix. The audio signals added (L+R), form the monaural signal which is the main channel. The subtracted signals (L-R) are modulated on a 38 KHz carrier, to form the sub-channel. A balanced modulator is used; because the system requires that the carrier at 38 KHz will be suppressed, leaving only the modulated audio information. The 38 KHz oscillator is divided by 2 to produce the cohered 19 KHz pilot signal. Both the carrier and the pilot signal should be purely harmonics (sinusoidal), otherwise some undesirable (spurious - noise) signals may appear in the composite spectrum.

The three components of the stereo signal, i.e. the main channel, the sub channel and the pilot tone, are combined at the proper ratios (45%, 45%, 10%), forming the composite output.

II. THE HARDWARE – GENERATION OF CARRIER AND PILOT SIGNALS

Before the DDS era, producing “clean” carrier and pilot signals at 38 and 19 KHz respectively, considered to be a difficult task. An oscillator based on a crystal or a ceramic resonator, was often used. Since there are not many 38 KHz resonators available in the market, carrier and pilot signals often produced after some divisions (usually by 12 and 24) from a 455-456 KHz ceramic resonator. The dividers were digital circuits based on flip-flops and modulo-x counters and they produced pulsed signals rather than “clean” sinusoids. Some filters had to be used for suppressing the harmonics and producing the sinusoids. Unfortunately, the filters could not fully suppress harmonics and they also produced some phase shift (pilot tone was phase sifted in respect to the carrier). Harmonics induced undesirable noise (indermodulation products) and significantly degraded the composite stereo signal. The phase shifts also, made carrier regeneration and coherent detection of the sub-channel problematic at the receiver.

After 90s decade, many designers preferred to use an alternative approach for carrier and pilot generation. That approach based on using a microcontroller for producing the carrier rather using an ordinary oscillator. The pilot tone was still derived by using division by 2. That approach gives some flexibility on choosing the reference crystal, but microcontrollers and dividers produce pulsed (digital) signals and strict filtering was yet essential.

Fortunately, now (in 2014) we have DDS, which gives unlimited control over phase shift and the ability to produce clean (purely sinusoids) signals with great frequency accuracy and stability. Reference clock frequency (or crystal choice) is not very critical in a high resolution DDS and signal generation becomes simple, robust and completely accurate. Using a DDS also diminishes the necessity of using complex (high order) filtering.

Here’s a breakdown of the internal circuitry of a DDS device: its main components are a *phase accumulator*, a means of *phase-to-amplitude conversion* (often a sine look-up table), and a DAC. These blocks are represented in Figure 3.

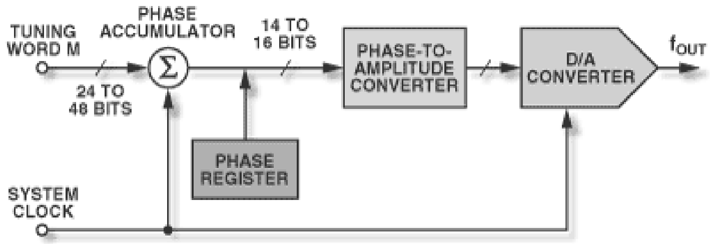


Fig. 3. *Components of a direct digital synthesizer.*

A DDS produces a sine wave at a given frequency. The frequency depends on two variables, the *reference-clock* frequency and the binary number programmed into the frequency register (*tuning word*).

The binary number in the frequency register provides the main input to the phase accumulator. If a sine look-up table is used, the phase accumulator computes a phase (angle) address for the look-up table, which outputs the digital value of amplitude—corresponding to the sine of that phase angle—to the DAC. The DAC, in turn, converts that number to a corresponding value of analog voltage or current. To generate a fixed-frequency sine wave, a constant value (the phase increment—which is determined by the binary number) is added to the phase accumulator with each clock cycle. If the phase increment is large, the phase accumulator will step quickly through the sine look-up table and thus generate a high frequency sine wave. If the phase increment is small, the phase accumulator will take many more steps, accordingly generating a slower waveform.

A phase-to-amplitude lookup table is used to convert the phase-accumulator’s instantaneous output value with unneeded less-significant bits eliminated by truncation into the sine-wave amplitude information that is presented to the (10-bit) D/A converter. The DDS architecture exploits the symmetrical nature of a sine wave and utilizes mapping logic to synthesize a complete sine wave from one-quarter-cycle of data from the phase accumulator. The phase-to- amplitude lookup table generates the remaining data by reading forward then back through the lookup table. This is shown pictorially in Figure 4.

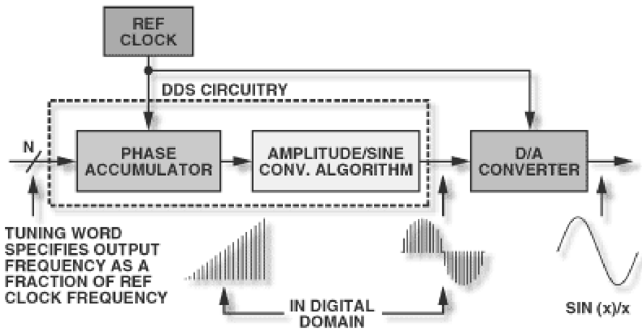
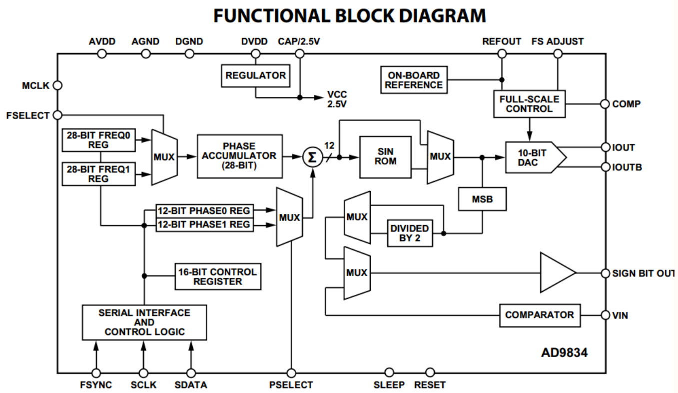


Fig.4 *Signal flow through the DDS architecture.*

A. The DDS Generator

In this fm-stereo encoder, we use Direct Digital Synthesis (DDS) for carrier and pilot tone generation. Referring to the DDS generator circuit section, the carrier and the pilot signal are generated from two AD9834 DDS ICs. Every AD9834 is used to generate a pure sinusoid signal. Both DDS IC's are kept synchronized by using the same reference clock, and their phase relationship can be digitally controlled. An 18F1220 PIC microcontroller is used to control the DDS generators threw I2C signalling interface. The I2C interface is implemented as "bit-banging" on normal I/O.



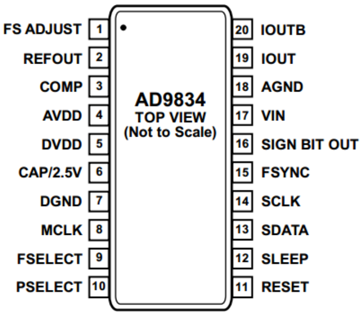


Photo 1. *The DDS generator. The carrier and the pilot signal are generated from two AD9834 DDS IC's. An 18F1220 PIC microcontroller (at the center of the photo) is used to control the DDS generators. Both DDS ICs are kept synchronized by the same reference clock (seen at the left side of the photo).*

*– **Fig. 6.** Functional Block Diagram AD9834 IC – **Fig.7** . Pin Configuration AD9834 IC.*

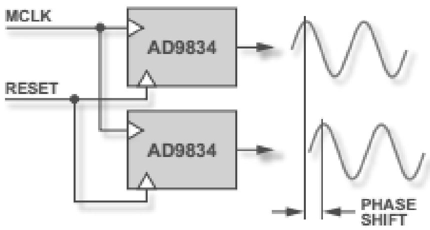


Fig.8 *Multiple DDS AD9834s IC's Synchronous Mode Setup of DDS generation with the same reset pin and master reference clock*

A reset, after power-up and prior to transferring any data to the DDS, sets the DDS output to a known phase, which serves as the common reference point that allows synchronization of multiple DDS devices. When new data is sent simultaneously to multiple DDS units, a coherent phase relationship can be maintained, and their relative phase offset can be predictably shifted by means of the phase-offset register.

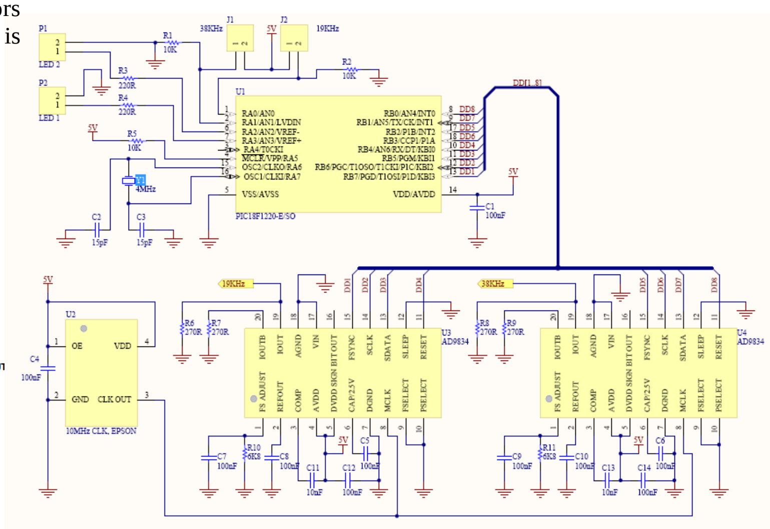


Fig. 5. *DDS Section – Carrier and Pilot Tone Generation*

The microcontroller is used to initiate the generators with the proper frequency and initial phase during start-up. It is also used to turn off or turn-on any generator at any moment, according to users will. User's commands are triggered from 2 external switches (J1 and J2). The AD9834 offers 28bits resolution over frequency and 12bits over phase control. By using a 10 MHz reference clock, we achieve frequency and phase accuracy of about 0.037 Hz (10MHz/2^28) and 0.09 degrees (360/2^12), respectively. The reference clock frequency is intentionally chosen to be high enough in order to can be easily filtered out from the carrier and the pilot signals, using only some simple R-C filters.

The source code is very simple. The microcontroller is used to initialize the DDS generators and then periodically checks J1 and J2, running on an infinitive loop. J1 and J2 are used to turn on or off the carrier and (or) the pilot signal, thus enabling or disabling the stereo broadcasting.

Besides main, there are only very few other routines in the code. These routines are responsible for initializing and turning on or off the carrier and (or) the pilot signal according to user will and also implementing the I2C interface, for the DDS chips, as "bit-banging" on normal I/O. Finally, there is also another essential parameter, regarding the correct phase relationship between the carrier and the pilot signal. The correct phase relationship between those signals is essential for achieving maximum "stereo-separation". The optimum phase relationship has been adjusted once threw code, and the stereo encoder was initially calibrated. Initial calibration constants are kept on some code lines (marked by the "Phase shift value" comment). These code lines are located in the void Pilot_on (void) routine and are used to set the initial phase parameter on the pilot tone DDS generator (please, refer to the AD9834's datasheet for more details about the phase parameter).

B. The Balanced Modulator

Modern approach on building a low frequency balanced modulator tends to be the use of DSP. However, traditional analogue techniques are still used due to simplicity. After all, the composite fm-stereo signal is a completely analogue signal. We may live in the digital era, but we still using the old and good analogue fm-stereo.

Following the tradition, we use an analogue balanced modulator for the generation of the 38 KHz sub-channel. The modulator is based on the well known MC1496 IC, which is able to suppress the carrier for more than 60dbs.

Photo 2*. The modulator is based on the well known MC1496 IC, which is able to suppress the carrier for more than 60dbs*

Carrier suppression is defined as the ratio of each sideband output to carrier output for the carrier and signal voltage levels specified. The carrier suppression for the MC1496, is very dependent on the carrier input level. A low value of the carrier results in lower signal gain, hence lower carrier suppression. A higher than optimum carrier level results in unnecessary device and circuit carrier feed through, which again degrades the suppression figure. The optimum carrier level for optimum carrier suppression at carrier frequencies in the vicinity of 50 kHz, is about 60mVrms (170 mVp-p). This Optimum value is achieved threw R47 adjustment.

Besides the carrier input, there is also another input for the L-R audio channel. The balanced modulator accepts both signals and performs the multiplication (L-R)*carrier in the time domain. A multiplication in the time domain is equivalent to frequency shifting in the frequency domain i.e. the L-R audio signal bandwidth is frequency shifted by the carrier frequency. This operation is better known as frequency mixing or shifting and the product of mixing is a DSB (Double Sided Band) signal.

There is a simple R-C filter at the carrier input of the modulator. This filter consists of the R56 and C48 and it is used to suppress the reference clock frequency (10 MHz). The DDS generates the carrier signal by using a 10bit DAC and the reference clock frequency is actually the sampling-frequency of the generated carrier signal. Since the reference clock frequency is much higher than the carrier frequency, it can be easily removed from the carrier signal by using a very simple low-pass (1st order) filter. The simple low-pass filter produces some phase shift, which is cancelled, threw appropriate phase shifting of the DDS generator. (see Fig. 8)

While the R47 is used to adjust carrier level at the input of the modulator, the R51 potentiometer is used to adjust the carrier suppression level. Carrier suppression better than 60db, can be easily achieved threw the appropriate adjustment of R51. For best performance, the modulator is powered from two independent voltage sources; +12 and -8V, respectively. These are the recommended supply voltages, as described in the MC1496 datasheet.

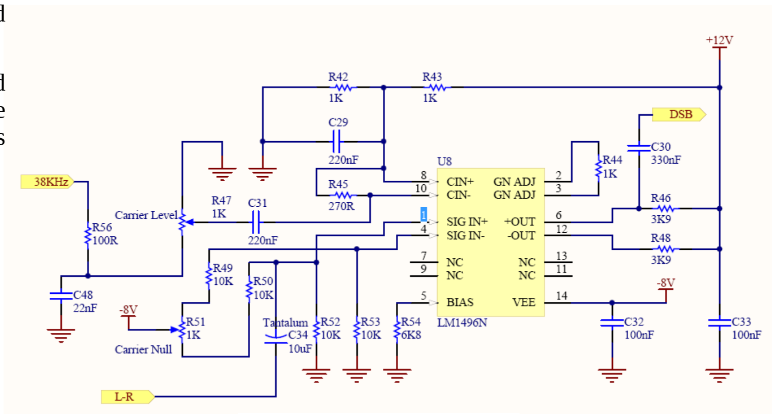


Fig. 9*. Balanced Modulator Section*

C. The Op-Amp Matrix

The heart of the fm-stereo generator is the matrix circuit. This circuit accepts the left and the right audio signals, the pilot tone and the DSBSC signal from the modulator, and performs the appropriate additions and subtractions, in order to produce the composite FM–stereo signal. The circuit also pre-emphasizes the left and right audio channel, just as normal monaural signal would be. The matrix circuit is based on operational amplifiers.

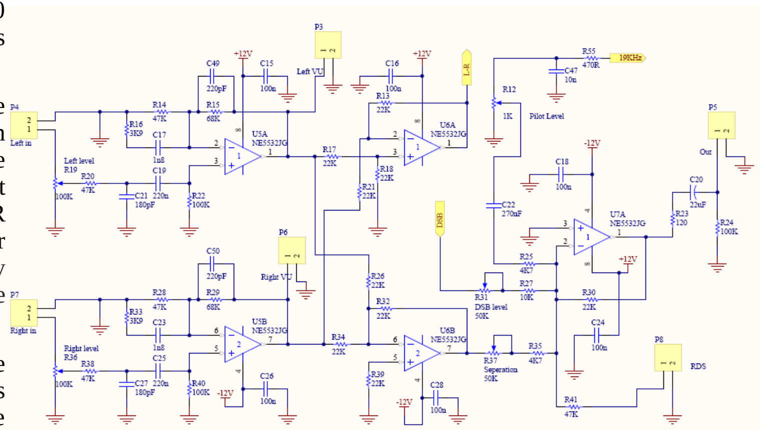


Fig. 10. *The matrix circuit is based on operational amplifiers*

Referring to the matrix electronic schematic, U5A and U5B are used to pre-emphasize the left and right audio channel. U5A, R14-16,R20, R22,C49, C19, C21 and U5B, R28-29, R33, R38, R40, C23, C27, C50 form pre-emphasis networks for the pre-emphasis of the left and the right audio channel, respectively. A pre-emphasis network is actually a high pass filter and pre-emphasis refers to a process designed to increase the magnitude of some higher frequencies with respect to the magnitude of lower frequencies. The pre-emphasis network characteristics are shown on figure 11.

In Europe, fm broadcasters use 50μς pre-emphasis, while it is 75μs in the U.S. Our FM-stereo generator prototype uses 50us pre-emphasis, because it was built and tested in Europe (Greece). However, it can be easily changed to 75μs by simply changing C17and C23 to 2.7nF.

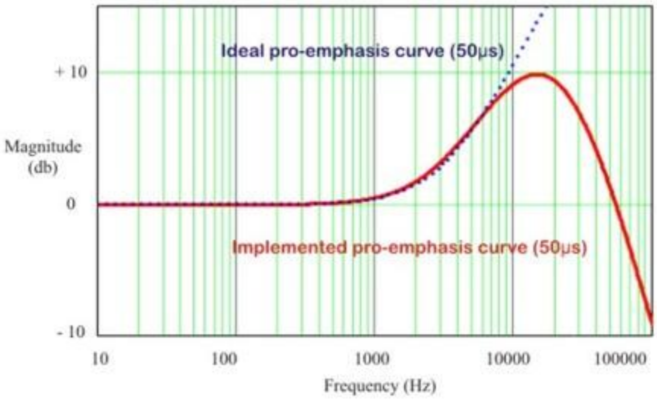


Fig. 11. *Pre-emphasis network response curve.*

Pre- emphasis on the transmitter and the minor operation (de-emphasis) on the receiver, are used to improve the overall signal-to-noise ratio by minimizing the adverse effects of the noise which is louder at higher frequencies. While the mirror operation is called de-emphasis, the system as a whole is called emphasis.

Fm channel is inherently very noisy and this makes emphasis very essential. Emphasis is also used in monaural broadcasting but it is even more important for FM-stereo. This is due to the fact that the fm-stereo signal carries most of its information in high frequencies located between 22 and 54 KHz and noise tends to be louder on those high frequencies. In the receiver side, decoding the stereo channel into left and right means that the noise is shifted down into the audible range.

Referring to the electronic schematic of the matrix again, U6A is used as a subtracter and produces the L-R signal, and U6B is used as an adder which produces the L+R sum. U7 is the final adder which accepts the pilot tone, the main channel and the sub-channel and produces the composite output. At this final stage, an additional input (P8) is providing for adding any SCA or RDS/RBDS subcarriers.

R12, R31 and R37 are used to adjust the proper ratios for combining the three components of the stereo signal, i.e. the pilot tone level, the sub channel level and the main channel level, respectively. Proper adjustment of these potentiometers is essential for the optimum operation of the stereo-encoder.

R55 and C47 are forming a low -pass filter for the pilot tone. This filter is used to eliminate the reference clock frequency (10 MHz), from the pilot signal. Besides the final output, which is P5, there are two other outputs. Those are the P3 and P6 outputs that are used to provide the left and the right audio signal, respectively, to an external VU-meter.

D. The Power Supply Unit

The fm-stereo generator uses a simple linear power supply unit which is based on 78XX and 79XX linear regulators.

Referring to the power supply electronic schematic, U9, U10, U11 and U12 are used to provide +5V, +12V, -12V and -8V respectively. The DDS generator section is powered from +5V only, while the modulator uses both +12V and -8V. The matrix section uses ±12V of symmetrical power supply.

III. Assembly Details

The prototype uses a double-sided printed circuit board with metal-plated holes. Excluding the AD9834 ICs, the PIC microcontroller and the clock generator, all other components are of through-hole type and they are placed on the top-side of the board. The microcontroller, the DDS ICs and the clock generator are placed on the bottom surface of the PCB. All resistors, except for those used on the matrix, are of 1/4W -5% type. In the matrix, I use low-tolerance 1% resistors and low tolerance (5%) capacitors.

The PIC microcontroller was programmed on board, using a MPLAB ICD 3 programmer from Microchip .

IV. Calibrating the FM-Stereo Encoder

The FM-generator, needs to be calibrated before use. The calibration process includes 5 steps as described below:

- ***Adjust the carrier level at the input of the modulator.*** Connect your oscilloscope on R47's tap. You should measure a 38 KHz sinus waveform, which is the carrier. Adjust R47, in order to get about 160mVp-p on its tap, in respect to ground.
- ***Achieve carrier null by means of the bias trim potentiometer R51.*** Turn R19 and R36 at zero scale (fully anticlockwise). Connect the oscilloscope on any pin of C30. Normally, you will get a 38 KHz sinus waveform on your oscilloscope. Adjust R51 in order to get 0Vp-p (null the carrier). Well, you will never get the absolute zero, but just some mVp-p (around 5mVp-p or less).
- ***Combine main-channel and sub-channel at the proper ratio.*** Set R36 at full-scale and R19 at zero-scale. Connect an audio signal generator on R audio input and apply a 1 KHz audio tone of about 0.6Vp-p. Short-circuit J2 to turn off the pilot tone. Measure the output of the generator using an oscilloscope. Adjust R37 and R31 in order to get a 3Vp-p signal, like the one shown on figure 12.

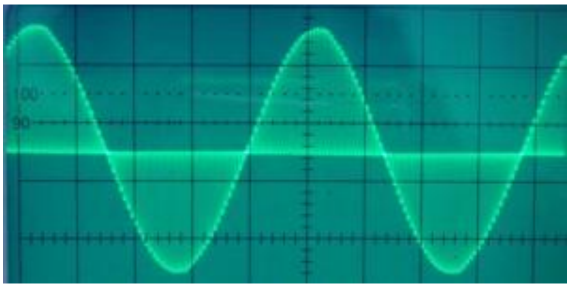


Fig. 12. *Right Channel only: Used to Balance Bain and Sub-channel*

- ***Adjust the pilot level. Set R19 and R36 at zero scale (full anticlockwise).*** Open J2 to turn on the pilot tone. Measure the output of the generator using an oscilloscope. You should measure a 19 KHz sine wave. Adjust R12 trimmer, in order to get about a 320mVp-p signal.
- ***Adjust the VU-meter.*** Set the left and right channel of the VU meter at full scale for 1Vp-p input. Adjust by using the trim potentiometers on VU-meter's board.

V. CONCLUSION

In this hybrid FM-stereo generator we use mixed digital and analog techniques in order to achieve optimum performance. We use Direct Digital Synthesis (DDS) to produce clean (purely sinusoids) signals with great frequency accuracy and stability for carrier and pilot tone generation . Reference clock frequency (or crystal choice) is not very critical in a high resolution DDS and signal generation becomes simple, robust and completely accurate. Using a DDS also diminishes the necessity of using complex (high order) filtering and we use very simple low-pass, 1st order filtering. The simple low-pass filter produces some phase shift, which is cancelled, threw appropriate phase shifting of the DDS generators. The correct phase relationship between the carrier (38khz) and the pilot (19khz) tone is essential for achieving maximum stereo-separation, and the optimum phase relationship has been adjusted once, threw code, according to trial and error method.

ACKNOWLEDGMENT

This project is a part of the production line of a new "Start – Up" small Business, "CircuitLib – Electronics". CircuitLib – Electronics was awarded with the First Prise at the "Secondary Student Contest on Business Plans and Innovative Ideas". The Student Contest on Business Plans and Innovative Ideas was organised from TEI of Crete, started on June of 2013 and ended on March of 2014.

REFERENCES

[1] Clifford B. Schrock, "FM Broadcast Measurements Using the Spectrum Analyzer"Application Note 26AX-3582-3 ,Techtronix 1981

[2] Eva Murphy,Colm Slattery "Direct Digital Synthesis (DDS) Controls Waveforms in Test, Measurement, and Communications"Analog Dialogue39-08,August(2005)

[3] PIC18F1220/1320 Data Sheet 18/20/28-Pin High-Performance, Enhanced Flash Microcontrollers with 10-Bit A/D and nanoWatt Technology 2007 Microchip Technology Inc.

[4] MC1496,MC1496B Balanced Modulators/Demodulators Datasheet On Semiconductor Components Industries, LLC,October2006 ,− Rev. 10

[5] Eva Murphy, Colm Slattery "All About Direct Digital Synthesis" Analog Dialogue 38-08, August (2004) http://www.analog.com/library/analogDialogue /

[6] Data Sheet AD9834 - 20 mW Power, 2.3 V to 5.5 V,75 MHz Complete DDS, Analog Devices www.analog.com

[7] CircuitLib – The Electronics Circuit Library www.circuitlib.com

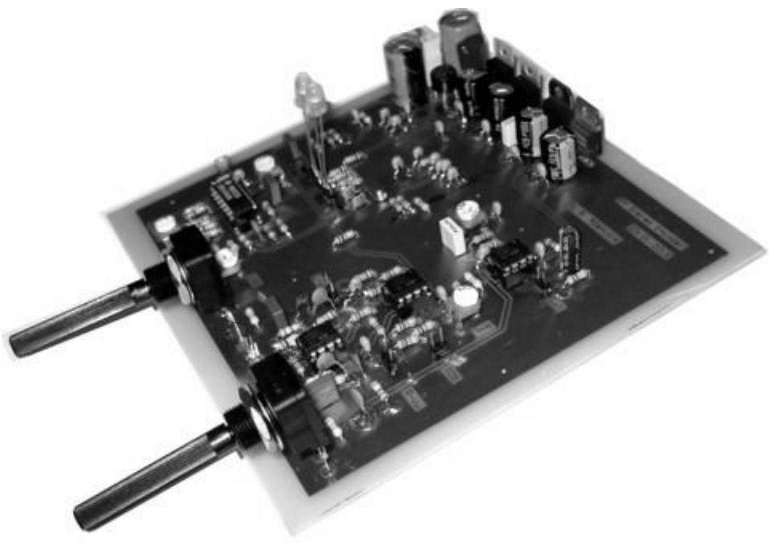


Photo3. *The electronic board of the hybrid Fm Stereo Encoder*

Manolis G. Tambouratzis was born in 1986 at Heraklion ,Crete Greece. In 2011 he graduated from the Department of Electronic Engineering, Faculty of Applied Sciences , TEI of Crete with grade " Very Good" 7.32 / 10 drawing up his internship at the Technical Department of the Hellenic Telecommunications Organisation (OTE). He holds a professional License "Radioelectronics Cat B" and a degree in Electronic Communications - Radio-Amateurs ' Cat 1 ' with the distinctive code "SV9RGJ" from 2012. His interests are focus on Electronic Circuits Technology for Telecommunication Applications (RF Systems Design), Wireless Telecommunications Systems Technology and Electromagnetic Fields Measurements from Base Stations Transmitters . He is a member of scientific societies IEEE (Student Branch - TEIoC), FITCE and EETEM and has attended numerous training seminars in the fields of Communications & Information Technology in recent years.

George A. Adamidis - Physicist and Electronic Engineer.
Studies: Physics (2000), MSc in Electronic Physics (2002), from Aristotle University of Thessaloniki, Greece.
Current occupation: High – School teacher in Physics & Electronics, Assistant Researcher in the Centre of Technological Research of Crete.
Research interests: Embedded Systems, RF technology, Smart Antennas, Antenna design and Electromagnetic Compatibility.