THE WEST RAND AMATEUR RADIO CLUB

December 2004
Volume 5, Issue 5

ANODE

Inside this issue:

Editor's Comments

December 2004
Volume 5 Issue 05

(PC version) Happy Holidays!
Happy Christmas/ Hanukah?

Qrp Power Tests Announced

Sunday November 21st, , December 26th and January 16th are the dates announced of the second Belgian U-B-A 'Candlelight Tests." In these tests, several stations transmit simultaneously on 80, 40 or 30 meters running micro power levels ranging from 100 milliwatts down to 5 milliwatts. The challenge for participating amateurs is to copy the code words embedded in the transmitted messages. Full information is on the U-B-A website. A link to it appears at www.uba.be/actual/candlelight/candlelight_en.html.

(GB2RS)

More Than Sixty Years Ago

The first London newspaper to receive news by wireless was the Daily Mail, taking a message from Marconi's at Chelmsford, on May 28, 1920. The June issue of Wireless World carried an article on the Mail's station and took the opportu-

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9 MHz Digital SSB Modulator

By Nico Palermo
IV3NWV
e-mail:
iv3nwv@microtelecom.it

Special points of interest:

- Contact details on back page (updated)
- New email address for Anode and ZS6WR. See back page

1. SUMMARY

This paper presents a digital single sideband modulator with a 9 MHz output. The core of the modulator, a numeric version of the Weaver modulator, fits into a Xilinx XC2S30 field programmable gate array. The quality of the single sideband output signal is superior to that of conventional analogue exciters. The carrier suppression and the unwanted sideband rejection are very high (about 80 dBpep) and spurious responses are better than 75 dBpep in the 0-18 MHz frequency range.

The modulator is equipped with an internal two-tone generator, helpful for the measurement of intermodulation products in the transmitter conversion/amplification chain, with a key input for CW operations, and with a secondary output which provides a 8.9985/9.0015 MHz beat carrier for SSB/CW product detectors.

2. INTRODUCTION

With the conventional numeric approach the synthesis of high frequency modulated signal is based on a DSP microcontroller which generates an intermediate signal, usually with a very low (or null) carrier frequency, and on a up converter, either analogue or digital, which shifts the sig-

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nity to do a bit of crystal gazing.

"The Daily Mail" installation consists chiefly of a six foot frame aerial of the solenoid type, wound with 48 turns of wire, used in conjunction with Marconi 7 valve high frequency amplifiers and detectors. Types 55A and 55D, which have been previously described in our pages and are familiar to most of our readers. Type 55 is one of the most sensitive receivers in existence and is particularly suitable for use with a loop aerial. The tuning arrangements permit of reception on wavelengths of from 600 metres to 18,000 metres. Damped and undamped waves and wireless speech can be equally well received on this apparatus, which is no amateur set but an instrument which has been thoroughly proven both in war and commerce, and is capable of detecting signals from any high power station within a radius of 3,000 miles. In a vision of the future one sees the inside of a newspaper office, where reporters are busy receiving "copy" from their colleagues in provincial towns, whilst automatic receivers click out tape records of news messages sent at 100 words a minute from the world's high power news distributing stations. From this to direct type setting by wireless is, maybe, not so far a cry as from Marconi's early experiments to his first great achievement, transatlantic wireless telegraphy!

"If, in addition, this future newspaper draws its electrical power from some huge Wireless Power Station, why then we shall have really begun in earnest to use that incomparable, universal medium, the aether.

"A visit to Carmelite House and a conversation with Daily Mail officials revealed that the latter intend to lose no time in assisting wireless and journalism to join hands. They look forward to the time When a reporter shall start for the scene of his "story" in an aeroplane "and arrive," one of them humorously interpolates and deliver his "copy" to headquarters by a system of linked wired and wireless telephony, the message being received at the paper's own wireless station. They intend to make as much use of Wireless as possible and entertain no doubt but that present day apparatus can fulfil all the demands likely to be laid upon it by Fleet Street in general. The idea of an "exclusive", message being flung out on an indiscriminating, generous aether, and intercepted by rival papers, created a disturbing ripple in the flow of conversation. Knowing that a similar objection has been levelled at wireless telegraphy for twenty years we do not view this question in quite such a serious light. There is this point, too, which must be taken into account directive wireless is probably not far distant.'

A very HAPPY Christmas

From: "KØHB"
<groupk0hb@earthlink.net>
Subject: SHE CUE, SHE CUE, OH &*^%$$#@, HIC!
Date: 2004-12-07 08:08

Dear Radio Friends,

It becomes my painful duty to write you a letter of apology instead of sending you the Christmas remembrance that I had intended for you.

Knowing so well the appetites, likes and dislikes of my close rrap friends, I made up a list of twelve of my best friends and went down and bought twelve quarts of "Royal Canadian" seven-year-old whiskey. Nothing but the very finest for my friends. I took this home and put it in my closet and intended to put it up in nice Christmas packages and send it out to you just before Christmas.

My wife, K0CKB, got to fumbling around in the closet and found the liquor and didn't understand just what I intended to do with it. So she came up and told me to empty the contents of every bottle down the sink, or else. So I proceeded with the unpleasant task.

I withdrew the cork from the first bottle and poured the contents down the sink, with the exception of one glass which I drank. I extracted the cork of the second bottle and

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likewise poured the contents down the sink, with the exception of the glass which I drank. I pulled the sink from the next bottle from the cork of the next and drank one sink out of it and threw the rest down the glass. I pulled the sink out of the next glass and poured the cork from the bottle. Then I corked the sink with the glass, bottled the drink and drank the pour.

When I had everything emptied, I steadied the house with one hand and counted the bottles, corks, glasses, and sinks with the other, which were twenty-nine. And, as the houses came by, I counted them again and finally had all the houses and bottles and corks and glasses and sinks counted, except one house which I drank.

So this accounts, dear friends, for your not receiving a more stimulating Christmas present than this letter. I sincerely hope that from some other source you get all the joys and happiness that come from Christmas remembrances.

"May the Lord take care of you, but not too soon."
73, Hans, K0HB, & Colleen, K0CKB

Final G5RV Logbook Retrieved From EBay

Meantime, word from the United Kingdom that the final log book of one of the worlds best-known radio amateurs has been saved for posterity. This, after it was put up for auction on eBay.

The late Louis Varney, G5RV, of G5RV antenna fame was one of the founding members of the Chelmsford Amateur Radio Society in 1936. So when Chelmsford club member Duncan Munro, M0KGK spotted that G5RV's last ever logbook was being auctioned, the club decided to buy it.

Munro did the bidding. Despite fierce competition he managed to secure the logbook in the last eight seconds of auction.

The logbook shows that G5RV's final QSO's took place on the 11th January 2000 and were , appropriately enough recorded on page 73 of the logbook. The stations contacted were Ron Glover, G0WGP, in West Sussex and club President Harry Heap, G5HF, in Chelmsford. Louis was, of course, using his famed G5RV antenna for all of the contacts. (GB2RS)

Uk Hams On The Air With Expanded 40 Meter Privileges

And finally this week, hams in the U-K have taken to the airwaves with their recently expanded privileges. RSGB Newsreader Jeramy Boot, G4NJH, is where it all took place:

UK radio amateurs gained access to new frequencies between 7.1 and 7.2MHz from 31st October. After dark, the band between 7.1 and 7.2MHz is still full of powerful broadcast stations, although some clear frequencies can be found.

The band became available on the second day of the CQ World Wide DX Phone Contest. Many UK stations took advantage of the new band by making contest QSOs with stations in the USA on their own frequency, instead of having to work 'split' as had been the case when contacting North America on 40 metres SSB.

During the daytime, numerous UK stations are now able to have virtually interference-free inter-G contacts above 7.1mhz.

The new band is available to radio amateurs on a Secondary, non-interference, basis until 2009. After the end of March 2009, the broadcast stations should move to other frequencies and 7.1 to 7.2MHz will then become an amateur Primary band.

Jeremy Boot, G4NJH, in Nottingham.

Ao-51 E-mail Request Box

AMSAT North America has announced the establishment of an E-mail address for AO-51 users to submit requests and ideas for Experimenters

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Wednesday operations. AMSAT says that this is a chance for the ham community that uses AO-51 to request the modes that they are interested in* seeing operated on during the weekly Wednesday times. All input will be noted but do not expect a reply to your E-mail. Send your requests or ideas to ao51-modes@amsat.org (AMSAT-NA)

[*hmm...kind of / like sending spam to cyber-space ed.*]

Ham Radio Licensing Stops And Starts Due To Computer Glitch

Ham radio licensing came to a screeching halt on November 5th. This after a software glitch caused the computer that issues licenses to do some very strange things. Mark bramovich, NT3V, has been following the developments from Philadelphia and gas more:

The FCC isn't offering any official comment on exactly what happened. But Amateur Radio Newsline has learned that the problems with the Universal Licensing System computer started after an Oct. 28 software upgrade installed by a government contractor.

The problems that followed prompted the FCC to order a five-day shutdown of the system which issues Amateur Radio licenses. The system came back on-line on Nov. 10. However, as of our deadline for this week's program, the FCC still planned a weekend shutdown of the U-L-S for maintenance and expected it to return to operation on Nov. 15.

The American Radio Relay League's Volunteer Examiner Coordinator Bart Jahnke, W9JJ, tells Amateur Radio Newsline he was among the first to detect problems while doing a routine check up on the application of a friend near the end of last month.

Jahnke says he discovered the application was - as he describes it - flagged or marked pending by the FCC for a review by a live person. He says the automated system and its software are set up for minimal human intervention. Jahnke says such a classification immediately raised concerns that something had gone wrong.

Jahnke tells me once he called his contact at the FCC to report the problem, he was told the computer contractor would be notified to investigate.

A couple days later, Jahnke says he was told the problem was fixed. But, as Jahnke tells me, he checked and found that the computer problem had gotten worse. He says he and other VECs were finding the U-L-S had begun issuing Group D call signs - the 2 X 3 ones - to new licensees out of sequence. He says only the first and third call-district applications somehow escaped getting caught up in the problem.

At that point, Jahnke tells Amateur Radio Newsline he and other VECs called the FCC. He says the commission told the contractor to pull the plug on the system on Nov. 5, run some diagnostics and fix it. A message posted on the FCC's website on Nov. 5 said simply: "The granting of Amateur applications has been temporarily suspended. We apologize for the inconvenience."

Prospective hams who had passed VE exams earlier and were awaiting their call sign were getting nervous. Some who received the out-of-sequence call signs were wondering whether the call signs were OK to use. A few had even received a second call sign, cancelling the first. In all, Jahnke says about 130 licensees were affected and thousands of license applications were backed up in the system.

By Nov. 10, Jahnke says the FCC began issuing letters cancelling the out-of-sequence call signs and awarding the proper call signs to new licensees. For some, he says it was their third call sign in a matter of days.

Jahnke says the FCC deserves credit for responding as fast as it did. He also has praise for the other VECs across the country

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Very High IP3 LNA for 144 MHz

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who were in touch the with FCC, too. Jahnke says he's crossing his fingers and hoping the planned weekend maintenance of the FCC U-L-S computer, won't undo the system fix.

For the Amateur Radio Newsline, I'm Mark Abramowicz, NT3V, in Philadelphia.

PL259/SO239 Connector Losses

[TenTec] Connector losses actually measured[TenTec] Connector losses actually measured from [Robert & Linda McGraw K4TAX]
To: tentec@contesting.com
Subject: [TenTec] Connector losses actually measured
From: Robert & Linda McGraw K4TAX <RMcGraw@Blomand.Net
Reply-to: tentec@contesting.com
Date: Mon, 10 May 2004 12:49:16 -0500
List-post: <mailto: tentec@contesting.com

You may have wondered as I have, about the so called UHF type of connector and how robust it really is with regard to its RF power handling abilities.

Here is some dialogue and test results that seemed to be of interest. Oh, sorry, I didn't mention Tentec, but for those of you running a Titan or Centurion, this may be of interest.

73
Bob, K4TAX

Regarding PL-259 loss, here is a prior post from K7FR who actually measured it.

Back in senior year at Washington State U (W7YH, Go Cougs!) we had to do a measurement project in Measurements Lab. Since there were two hams in the Lab we decided to measure losses in coax connectors (the Prof was a ham too).

We set up a calorimeter and measured I**2R losses from DC to 2 GHz for a PL259/SO239 combo (did it for BNC and N too...hey it was a senior project).

Here are some of the results from my Lab Notes:

Input power = 1,000 Watts (100V, 10A @ DC, homebrew 4-1000 .1-30 MHz, borrowed USAF signal source 30-2,000 MHz (black box from Fairchild AFB), Bird dummy load) (We used a kW because neither of us had ever run more than 100 Watts...power trip!)

f (MHz)	Loss (W)	dB
0.1	1	-0.00435
1	1.2	-0.00521
10	1.3	-0.00565
20	1.5	-0.00652
30	1.8	-0.00782
50	2.2	-0.00957
100	2.6	-0.01131
200	3.5	-0.01523
300	5	-0.02177
400	7	-0.03051
500	10	-0.04365
1000	15	-0.06564
1250	18	-0.07889
1500	28	-0.12334
1750	39	-0.17277**
2000	100	-0.45757**

** Connector failed before calorimeter stabilized.

We attributed the steep upswing after 100MHz to the finish on the connector, not the connector design. Nickel plating seems to exhibit non-linearity above 100MHz. The N and BNC runs were much better. BNC went flaky above 600MHZ (RG-58 size, RG-8 BNC went to 1000 MHz). We were able to isolate cable loss from connector loss by building a Teflon box around the connector body and only "viewing" the inside of the box with the sensor.

The Department Chair was not at all happy that this Teflon box cost $750 to build (Teflon was rare in 1977). As you can see from the table we experienced two failures. Both were due to the solder melting in the probe part of the connector. The 1250 and 1500 Watt runs showed discoloration but no melting. The values for 1750 and 2000 MHz were the calculated values at the time of failure. Each run took 1 hour, these two failed 28 and 17 minutes into the test.

We experienced a failure of an N connector at 2000MHz. We ran the output up in 100 Watt steps until we observed a sharp up turn in losses. We were able to boil the water in the

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Very High IP3 LNA for 144 MHz

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calorimeter at 15000 Watts and at 17100 Watts the fingers inside the connector relaxed and started arcing.

Before this experiment I was paranoid about my connectors. Since then I have only been concerned with the quality of the assembly and water ingress.

My take on it.......

73 Gary K7FR

Good job!

Experience has always told me that .5 db per connector was silly, especially at anything below UHF. I think the figure arose from the folks in the high UHF and microwave business, where connector loss becomes something more than virtually immeasurable. I suspect the relatively high loss at you measured at 1 GHz is not all that much more than the loss in an equivalent piece of line.

The non-linearity of nickel plating is well established. In high quality land mobile installations where there are high power transmitters and very sensitive receivers sharing a common site, it is common to find nickel plated connectors banned. The non-linearity apparently arises from the fact that nickel is somewhat ferromagnetic. Our old friend the hysteresis curve.

I still use the ,5 db figure when I work out microwave link budgets. It's always better to predict a little more loss than is actually there.

Gray

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TenTec@contesting.com
http://lists.contesting.com/mailman/listinfo/tentec

Homebrew double sideband

From: "Tom Coates" <tecoates@comcast.net>
Subject: Re: Homebrew double sideband
Date: 2003-04-21 03:56

Have you considered writing an article about your rig? This magazine may be interested:

`http://www.njqrp.org/data/qrp_homebrewer.html`

What you've posted here could easily be adapted into an effective query letter to the editor.

Tom, N3IJ

"Bill Meara" <n2cqr@clix.pt> wrote in message news:97f305da.0304192236.5f9c82fc@posting.google.com...
With sunspot numbers declining I reluctantly decided to shift my portable one watt DSB HB transceiver from 17 meters to 20 meters.

This rig is a very simple DC RX with a corresponding transmitter side. VXO is the only common stage. On TX, the VXO feeds an NE602 balanced modulator. I used an LM386 as the mike amp (that's what I had available). For the microphone I use an old SONY walkman mike.

On RX, I have a band pass filter followed by an NE602 functioning as an RF amp. Balanced output to another NE602, this one the product detector. Then ANOTHER NE602 as an AF amp (again, balanced input) followed by an LM386. I get the NE602s to work as amps by putting five volts on pin 6. Nicest DC RX I've ever used. No common mode hum, even with AC supply.

Shifting from 17 to 20 was easy. Some 50 pf caps across the few tuned circuits brought them into the band. I got two new crystals from JAN (very nice folks) and rigged up a panel switch to go from one to the other. With the two rocks I can tune 14.185 --14.211.

I had some trouble with the carrier suppression, but I found that I could improve it by reducing the VXO energy going into the 602.

Instead of the recommended

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Very High IP3 LNA for 144 MHz

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200 milliVolts, I went to around 20 by using a "gimmick" cap (just two wires twisted together) between pin 6 and the VXO. By varying the amount of twist I could reduce the carrier to minimal levels. This gimmick was fun.

The rig is working very well. I'm getting 1-2 watts PEP out. In two days I've worked Spain, Italy, and the U.K. This morning I had a nice 'ragchew' with K3LP in Maryland. It is a real hoot to tell people that they can switch to LSB and still copy just fine.

I don't think 'QRPers' should be scared away by the big guns on 20 phone. And I think the simplicity of DSB makes it an ideal choice for home brewers of intermediate skill level who want to break into the world of phone.

If you hear me, please give me a shout. USB or LSB!

73 from the Azores
Bill CU2JL N2CQR
http://planeta.clix.pt/n2cqr

More on the Tayloe detector
Tayloe detector by Sakari Mattila, VK2XIN / OH2AZG
(sam@isd.canberra.edu.au)

Tayloe detector is a direct conversion detector, which appeared on several mailing lists in mid October, 1998. The inventor, Dan Tayloe, N7VE call is switched integrator. It bears close resemblance to switched capacitor filter.

The basic idea is to store average voltage of each four parts of a cycle of incoming RF into four capacitors. If the RF is pure carrier, constant voltages will be produced. Modulated RF produces slowly variable voltages, which corresponds I (phase) and Q (quadrature) signals in a quadrature detector. If the RF is audio modulated, I and Q signals will be audio signals.

It is known, that all modulation types can be decoded from I and Q signals within detectors bandwidth. A stereo DSP processor will be needed for universal detector. Dan Tayloe used phasing type decoder for SSB signals.

The minimum detectable signal depends on audio amplifiers, like in any direct conversion receiver. The dynamic range is much better than in commercial receivers. Thus Tayloe detector seems to be good for measuring instruments.

The electronic switch must be fast enough to switch at four times the received frequency. Internal resistance of the switch must be small, much less than 50 ohms. The suggestion is 74CBT3253, which should work up to 100 - 150 MHz. The original detector did not have any RF amplifier, but I would add it, because there is high-level noise coming out of the switch at four times received frequency and harmonics.

This is partial quote from the original mailing list message:

MDS = Minimum Detectable Signal
2TDR = Two Tone Dynamic Range
3rd = Third order intercept

Radio	MDS dBm	2TDR dB	Blocking dBm	
Tayloe	-136	111	+30 (3rd)	< --- without RF amplifier
FT101E	-142	60		
TS520	-132	63		
R390	-136	82		
7553B	-146	88		<--- Collins 'S' line radio
Corsair	-128	90		
GQ40	-126	90	125	
NN1G	-132	90		
Nor. 40	-136	88	108	
Sierra	-132	88	100	
FT1000	-125	95	125	
TS930	-133	95		
Breed	-120	70	90	
Hayward	-128	95	125	<--- Hayward's criteria for contest
Mini R2	-136	96		<---- from QST article

Information for other radios is from Jim Duffy, KK6MC & Az ScQRPion, Kent Torell.

Dan Tayloe, N7VE, Phoenix, AZ, Az ScQRPions, QRPL # 696 was Quoted in the original message as the inven-

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tor of the Tayloe detector.

The message appeared on these mailing lists: bpsk@ qth. net; laser@ qth.net; rsgb_lf_group @ blacksheep. org;

Seasons Greetings to all

JB

9 MHz Digital SSB Modulator

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nal to the desired output frequency. My first all-mode digital HF transmitter (1991) was based on this architecture and used a Motorola DSP56001 together with the famous Analog Devices AD7008 direct digital synthesizer [1].

In the last years reconfigurable logic circuits have made possible the integration of more and more complex functions into a single integrated circuit, leading to small and cheap "Systems on Chip". Following the SoC approach, I developed a compact digital SSB exciter which fits into a small (30,000 gates) Xilinx FPGA, the XC2S30. The logic core includes all the functions required to transform a audio input signal into a 36 MSPS, 9 MHz SSB signal which is converted back in the analogue domain by a 14-bit DAC, an AD9754 (or by a cheaper, pin to pin equivalent, 10-bit DAC AD9760). The core integrates a 12 KSPS 10-bit sigma-delta audio ADC and a 14-bit numeric version of the classical Weaver modulator. Using serial arithmetic circuits and efficient frequency conversion/filtering algorithms, it has been possible to contain the core area to four hundred slices and three, 4,096-bits, RAM Blocks (approx. 90% of the slices and 50% of the Block RAM available in a XC2S30 FPGA).

3. DIGITAL SYNTHESIS OF A SINGLE SIDE BAND SIGNAL

3.1. The Weaver Modulator

The block diagram of an analogue Weaver modulator [2] is shown in fig.1. The input signal spectrum is converted to a zero i.f. band pass signal by means of a quadrature frequency converter and a low frequency oscillator. The zero i.f. signal is then filtered by two identical low pass filters which suppress the unwanted sideband and converted to the final frequency by a second high frequency quadrature upconverter.

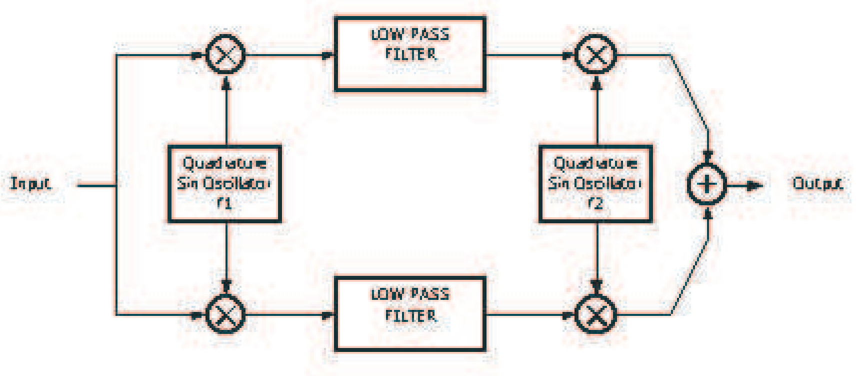

Fig.1 – The Weaver modulator

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9 MHz Digital SSB Modulator

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Fig.2 shows the processing steps operated on the spectrum of an audio input signal to obtain the desired IF SSB output. For a radio communication system it is widely accepted to limit the input audio bandwidth to 2400 Hz (from 300 to 2700 Hz). The audio input is a pass band signal with a 1500 Hz center frequency.

With the first mixing one of the two side bands of the input signal is converted to a zero carrier frequency signal (usually called by engineers a complex low pass signal, or zero i.f. signal). To obtain such a zero i.f. signal, the frequency f1 of the l.f. quadrature oscillator should equal that of the input signal, that's to say 1500 Hz.

The other sideband is moved (in absolute) to an higher frequency. It is up to the two low pass filters to reject it, while passing unaltered the zero i.f. sideband to the next conversion stage. The low pass filters should exhibit a low attenuation and ripple in the band -1200/1200 Hz (wanted sideband) and introduce a large attenuation for frequencies, in absolute, greater than 1800 Hz (unwanted sideband).

The second quadrature mixer converts the zero IF single sideband signal to the desired IF frequency f2. Fig.2 shows the generation of an USB signal. It is sufficient to invert the sign of one of the quadrature oscillator outputs to obtain a spectral inversion and the generation of an LSB signal.

Note that the suppression of the carrier is due to the action of the first two mixers, which are supposed to be perfectly balanced, and to the assumption that the input signal has no DC component.

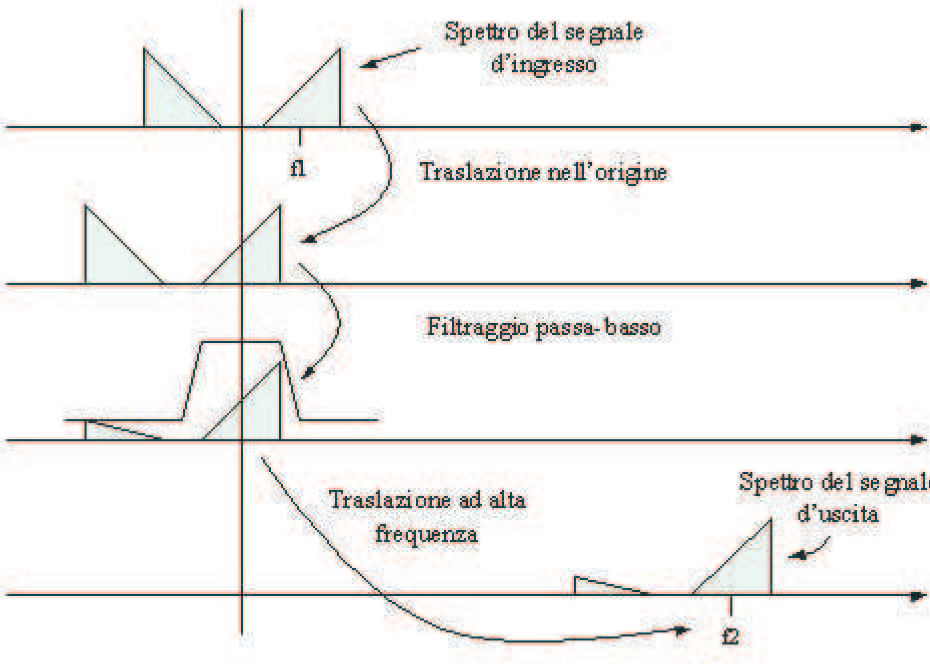

Fig. 2 - SSB signal generation with the Weaver modulator.

3.2. The Digital Version

From the point of view of the centre frequency of the processed signals, we can divide a SSB Weaver modulator with i.f. or r.f. output in two sections. The first one is an audio section which processes a low frequency input signal and is made by the first frequency converter with its 1500 Hz quadrature oscillator and the two low pass filters. The second one is an high frequency section and is formed essentially by an high frequency quadrature up converter. A numerical implementation of the modulator shall take in account the different processing needs of the audio and the h.f. sections.

In a real time DSP circuit, the complexity is proportional to the number of arithmetic operations per second required to perform the desired task.

Since the number of operations per second is the product of the number of operations required

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9 MHz Digital SSB Modulator

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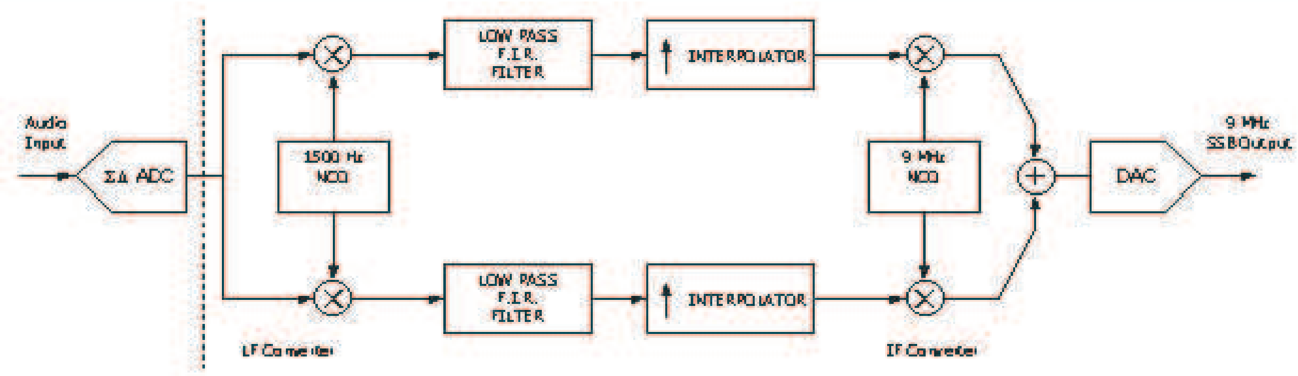
per sample times the sampling frequency, it is important both to reduce the number of operations/sample and minimize the sampling frequency. The reduction of the number of operations/sample is accomplished with the choice of the right algorithm. The analysis and the simulation of a particular DSP algorithm help to do the right balance between the circuit performance, which depends upon the algorithm itself and the calculus precision, and the required logic resources. But if we fail to reduce the sampling frequency to the minimum value which allows easy transformations of signals between the analogue and the digital domains, we are wasting processing resources by sure.

In a SSB Weaver modulator the input signal has a bandwidth of roughly 3 KHz and can be sampled at some ten kilohertz rate. The output is a h.f. signal instead and requires a much higher sampling rate, usually not less than three times the maximum output carrier frequency. A numerical version of the modulator is shown in fig. 3.

It can be noted that two circuits named "interpolators" are inserted between the leftmost l.f. section of the modulator, sampled at an audio rate, and the r.f. section (rightmost) which is sampled at a very high frequency. The function of the interpolators is then that of sample rate conversion, from a low value to an higher one.

Fig. 3 – Block diagram of the digital SSB Weaver modulator



The quality of a digitally synthesized SSB signal can be characterized by the same parameters used in analog modulators:
linearity and the in band (narrow band) SNR of the modulated signal, unwanted sideband rejection, carrier suppression and wide band spectral purity.

In band SNR depends upon the input sampling frequency, the audio AD converter precision and the quality of the l.f. frequency converter. Carrier suppression and unwanted side band rejection depend upon the offset of the AD converter and the frequency response of the low pass filters. The wide band spectral purity of the r.f. output depends upon the interpolators, the h.f. frequency converter, the output DA converter and output sampling frequency.

In this modulator the precision, that's to say the width of the signal buses, has been kept as large as possible.

In the audio section, sampled at 12 KSPS, the 1500 Hz down converter has a 16 bit precision. The MAC units used in the low pass FIR filters have 16 bit operands (data and coefficients) and 22 bit outputs, which are then truncated to 16 bit before reaching the interpolators.

In the high sampling rate section, each interpolator produce a 14 bit quantized output. Since each of them is a cascade of CIC filters, which essentially compute a moving average of its input, a larger precision (up to 28 bits) is required for the intermediate results before truncation. The sampling rate of the zero i.f. ssb signal is incre-
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9 MHz Digital SSB Modulator

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mented in three stages, from 12 KSPS to 180 KSPS by a 4th order, 32 bit, x15 serial interpolator, from 180 KSPS to 4.5 MSPS by a 2nd order, 16 bit, x25 parallel interpolator and from 4.5 MSPS to 36 MSPS by a 1st order, x8 parallel interpolator (which is simply a 14 bit hold register).

The 36 MSPS zero i.f. ssb signal is finally converted to a 9 MHz i. f. signal by a simple 14 bit Fs/4 up/down converter.

4. IMPLEMENTATION ON FPGA

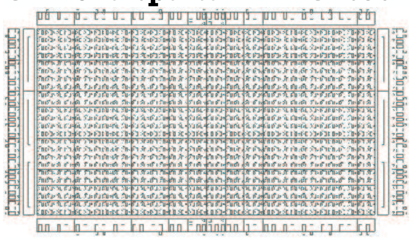
4.1. Architecture of a Spartan-II FPGA

In Xilinx's Spartan-II FPGAs, each cell (CLB, or Cell Logic Block) is made by two "slices". As shown in fig. 4, each slice contains two Flip-Flops, two look-up tables which can synthesize any combinatorial Boolean function with 4 inputs, and carry chain logic dedicated to arithmetic functions. Internal propagation times are in the nanosecond order of magnitude and the maximum toggle frequency in good pipelined designs can be much higher than 100 MHz.

Fig. 4 – Block diagram of a Spartan-II slice

The device XC2S30 contains an array of 12 x 18 CLBs (fig. 5), that's to say 432 slices (864 Flip-Flops) and is equipped with 6 blocks of 4,096 bit synchronous dual-ported RAM. Half of these RAM blocks have been used to store the samples and the coefficients of the modulator low pass FIR filters.

Fig. 5 – The array 12 x 18 CLB of a Spartan-II XC2S30



4.2. Sampling and AD conversion of the audio input.

The audio input is sampled at 12 KSPS with a 10 bit sigma-delta AD converter.

The converter has been realized with an operational amplifier which integrates the error between the input signal and its 1-bit quantized replica, and compare the error versus a fixed threshold, generating a pulse train which drives the AD converter logic. The pulse train is sampled at 6 MSPS, feedback to the operational amplifier and decimated to 12 KSPS by a cascade of two 1st order

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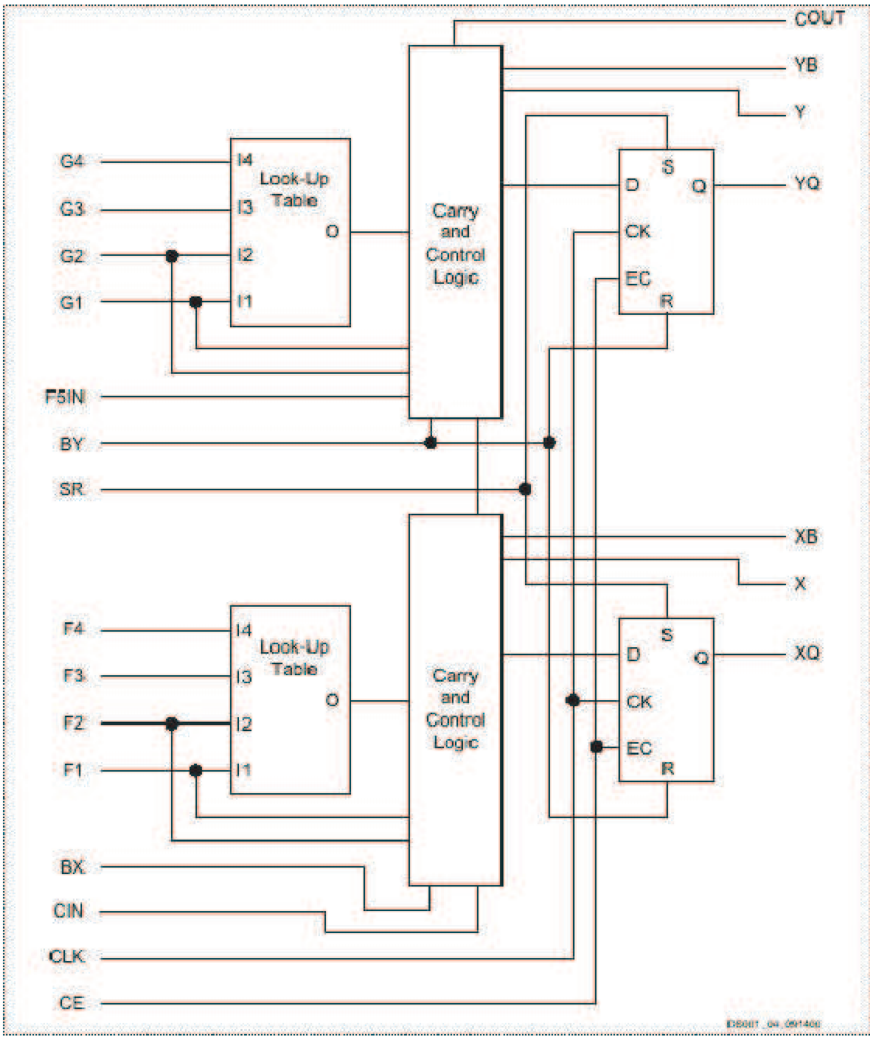

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x25 decimators which produce a 10 bit digital version of the input signal.

4.3. Frequency conversion of the digital audio.

The quadrature 1500 Hz down converter has been implemented with a CORDIC algorithm [3]. The COordinate Rotation Digital Calculator, ideated by J. Volder in 1956 [4][5], solves the problem of coordinate rotations with simple add/subtract/shift operations and, since a frequency conversion is just a rotation of a vector on a plane, is the preferred way to realize frequency converters in the hardware. In the CORDIC an arbitrary rotation of the input vector is obtained iteratively and by successive approximations of the rotation angle.

The general formulas that transform the coordinates of a vector (x,y) rotated by an angle phi into the new coordinates (x',y'):

```
x’ = cos(phi) x - sin(phi) y = cos(phi) ( x – tan(phi) y),
y’ = cos(phi) y + sin(phi) x = cos(phi) ( y + tan(phi) x)
```

may in fact be simplified if the rotation angle is such that tan(phi) = ±2-i.

In this case, omitting the multiplicative term cos(phi) which is constant and independent upon the rotation direction, the formulas become:

x’ = (x - ±2-i y),

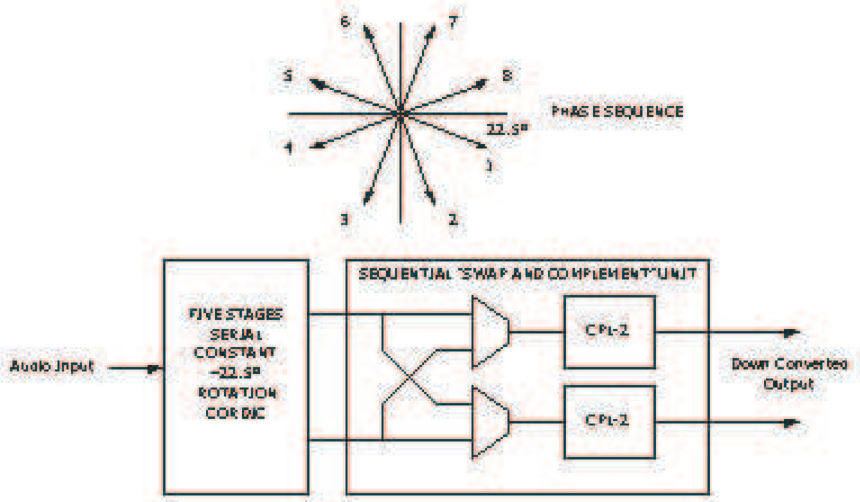
y’ = (y + ±2-i x)

and require only add, subtract and shift operations. An arbitrary rotation is then obtained by decomposition of the angle into a finite sum of angles whose values satisfy the rule tan(phi) = ±2-i and applying the above formulas iteratively.

The 1500 Hz frequency conversion of an audio signal sampled at 12 KHz is just a sequential and periodic rotation of the input audio signal by an angle of 1500/1200 · 360° = 45°.

In this ssb modulator, the frequency converter rotates the audio signal by a fixed angle phi = -22.5° by means of a 16 bit, 5 stages, serial CORDIC and uses the symmetry of the rotation sequence to produce the desired output signal, as shown in Fig. 6.

Fig. 6 – 1500 Hz Digital down conversion of the audio signal



The desired 22.5° rotation is approximated by the five term sum:

atan(2-1) - atan(2-4) - atan(2-7) - atan(2-10) + atan(2-12) = 22.499138...°,

with a very good accuracy and an high spectral purity, obtained in just 80 clock cycles/sample (16 cycles per stage) and a small fraction of logic resources which would be required to realize the same down converter with the classic multiplier/quadrature direct digital synthesis architecture.

4.4 Low pass filtering. Unwanted sideband rejection.

In the SSB Weaver modulator the suppression of the unwanted sideband equals the stop band attenuation of the

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low pass filters. To obtain a 50/60 dB rejection over a bandwidth which is about one fourth of the sampling frequency with a linear phase flat response in the pass band, FIR filters with at least 50 taps are required. Since the audio sampling frequency is a small fraction of the clock frequency there are a lot of clock cycles available to compute the filter output (in this modulator, 36 MHz/12 KSPS = 3000 clock cycles per sample) and the arithmetic operations required by the filters can be serialized in order to largely reduce the CLB requirements. DSP microcontrollers rely on one or more MAC units which execute in a single clock cycle and allow to compute the response of a FIR filter in as many clock cycles as the filter taps. To reject adequately the unwanted sideband we need just a fraction of such a computational speed. Each low pass filter has to compute 50 MACs at 12 KSPS, that's to say 600,000 MAC/s and a single cycle MAC unit is not required. Using a 36 MHz clock, there are 36,000,000 Hz / 600,000 MAC/s = 60 clock cycles available to compute (serially) a MAC. The FIR filters I developed use 16 x 16 bit input serial multipliers which execute in 16 clock cycles.

With a 36 MHz clock and a 12 KSPS processing rate the upper limit to the tap number of the filters is 185, almost four time the real requirements. I exploited this capacity to implement two 127 taps, 16 bit coefficients, symmetric Remez FIR filter with a 0.2 dB ripple in the pass band [0, 1200 Hz] and a 80 dB stop band [1500, 6000 Hz] attenuation. They not only reject the unwanted sideband but also the carrier which falls exactly at the lower limit of the stop band. The frequency response of the filters is shown in fig. 7.

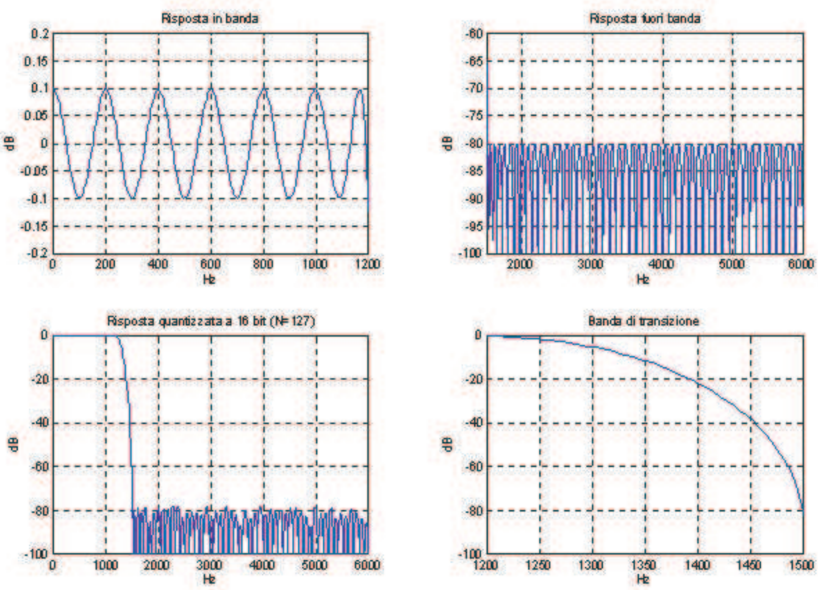

Fig. 7 – Frequency response of the sideband rejection filter

4.5. Interpolation to 36 MSPS

The spectrum of a sampled signal is periodic with period equal to the sampling frequency. The single sideband signal generated by the audio section of the Weaver modulator is sampled at 12 KSPS and its spectrum contains aliased images spaced 12 KHz. The interpolators have to resample the input signals at 36 MSPS and attenuate the unwanted images to an acceptable level. This result is obtained with filters which introduce a large attenuation at multiples of the input sampling frequency, as shown in the example of fig. 8.

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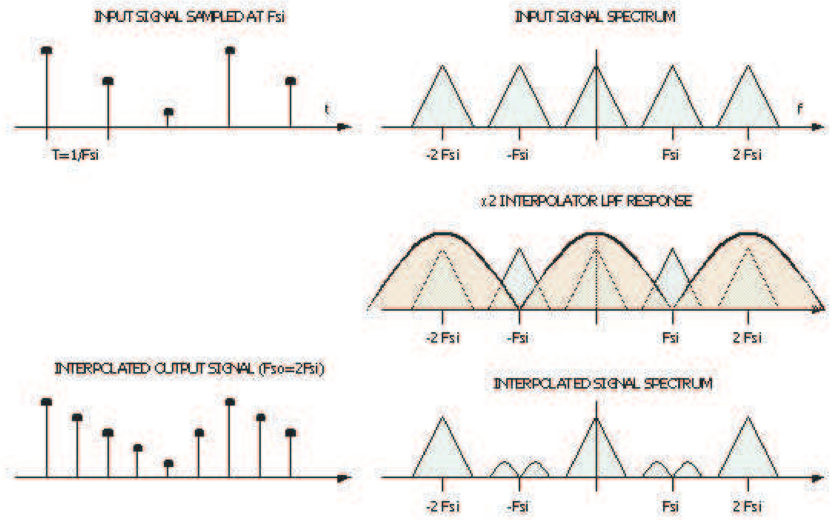

Fig. 8 – Interpolation of a sampled signal

A class of filters which is very efficient to the hardware implementation of interpolators (or decimators, if the sampling rate has to be lowered) is that of the CIC (Cascaded Integrator Comb) filters, due to E.B. Hogenauer [6]. Fig. 9 shows the schematic diagram of a decimator and an interpolator realized with CIC filters.

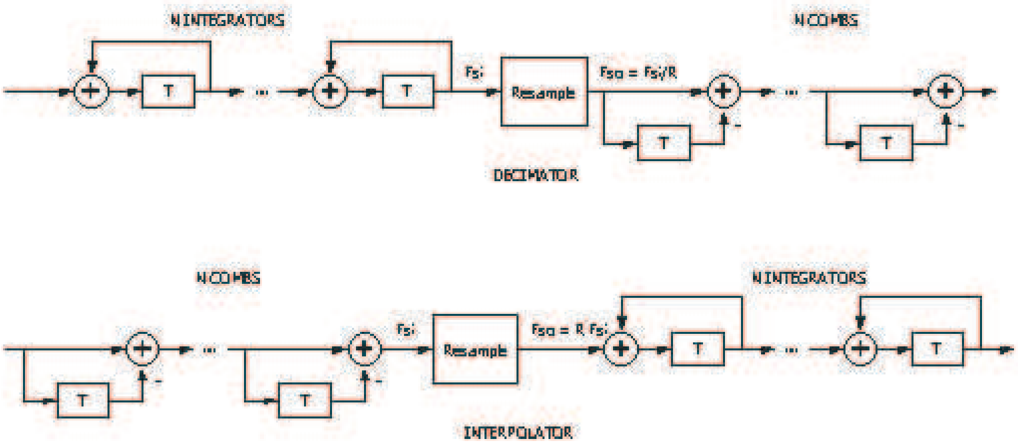

Fig. 9 – Schematic diagram of Nth order CIC decimators and interpolators

The frequency response of a CIC filter of order N, interpolation factor R and output sample frequency fso is:

The transfer function has Nth order transmission zeros at frequencies multiple of the input sampling frequency fsi = fso/R. The zeros are centred on the images and attenuate them considerably.

The interpolators used in the SSB modulator resample the l.f. signal by a factor R=3000 and attenuates the images by more than 70 dB. They are implemented with a cascade of three interpolating stages.

The first stage includes a 4th order x15 serial CIC filter with 16 bit input (R=4, N=15) and resample the input signal at 180 KSPS. Since its DC gain is RN-1 = 3375, it requires log2(3375) = 12 extra bits to avoid output overflow, leading to a 28 bit result which is truncated to 16 bits before entering the second stage. The second stage is a 16 bit, 2nd order, x25 parallel interpolator (N=2, R=25) and resample the output of the first stage to 4.5 MSPS. The third and last stage is a 14 bit, 1st order, x8 interpolator (N=1, R=8) and, since it has a rectangular impulse response, it doesn't consume any logic resource (it is just the accumulator register of the second stage).

4.6. Up conversion to 9 MHz.

A tuneable, high sample rate, frequency converter consumes a lot of logic. The CORDIC algorithm simplifies the conversion process but in the generation of h.f. signals it has to be necessarily implemented in parallel form. Each CORDIC stages requires two

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adders/subtracters for the rotation of the input vector and a third adder/subtracter for the calculus of the residual angle which feeds the next stages. For adequate spectral purity in transmission chains, the converter
can require 16 cascaded stages with 16 bits buses for the input complex vector and for the rotation angle. A pipelined version of such a converter would use 48 adders/subtracters with registered output, for a total of 768 Flip-Flops, that's to say almost all the logic of a XC2S30 device.

Anyway, if tuning is not required and the output frequency is limited to exactly one fourth of the sampling frequency, the converter can be realized with a much less logic circuits. This is the reason a 36 MSPS sample frequency has been chosen in this modulator. In fact in a digital frequency converter with Fs/4 output frequency, the input signal vector has to be rotated sequentially by angles which are multiples of 90°. This operation translates in a four cycle periodic and sequential exchange and two-complement of the input vector components, as shown in fig. 10.

On a Xc2S30 a 14 bit Fs/4 up-down parallel frequency converter fits into 8 slices (16 FFs) and runs at almost 200 MSPS.

5. CIRCUIT DESCRIPTION

The schematic diagram of the SSB modulator is shown here. The modulator requires a stabilized +5 Vdc power supply, provided at pin 10 of the JP3 connector.

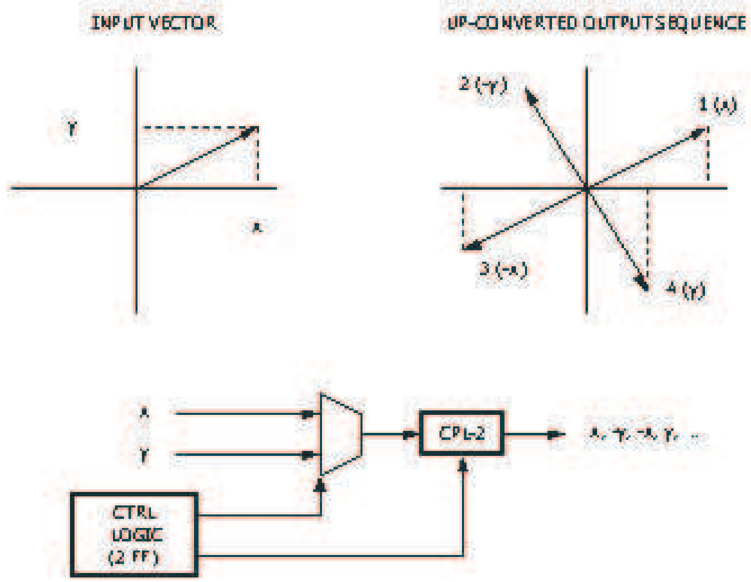

Fig. 10 – A fast Fs/4 frequency converter

The input audio signal (pin 1 of JP3) is filtered and amplified by the operational amplifier IC2A.

The input signal level required to drive the modulator is about 100 mVpp. The antiparallel couple of diodes BAV99 (D4) limits the gain of the preamplifier and avoid the saturation of the next stages. The section IC2B of the OPA2340UA op-amp works as the integrator/comparator section of the audio AD converter as explained in the paragraph 4.2. The inputs PTT\, BFO\, SSB/CW\, USB/LSB\ and TWOTONE\ available at connector JP3 control the operational mode of the modulator. PTT\ enables the audio input, BFO\ enables the generation of the beat signal for the receiver product detector, SSB/CW\ select the corresponding modulation, USB/LSB\ selects the transmitted side band, KEY\ is the CW key input, and TWOTONE\ enables the internal two tone generator.

All the input interface signals available on JP3 are active low and should be driven by open collector sources or switches connected to ground.

The programmable gate array IC3 (an XC2S30-5VQ100C) processes the audio signal and the CW key input to generate a digital, 14-bit, 9 MHz ssb output signal which is converted in the analogue domain by the DA converter IC4 (an AD9754, or a

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cheaper pin-to-pin compatible 10 bit AD9760 converter) and filtered by the anti-alias filters formed by L2-L6 and C32-C38. The modulator has two outputs, TXOUT and BFOOUT, available at connector JP4. The p.e.p. output level is -11 dBm p.e.p. (approx. 90 mVp on a 50 ohm termination).

IC1 is a 512 KBit serial EEPROM which stores the FPGA configuration bitstream.

The bit stream is uploaded in a fraction of second when the the power supply is applied to the SSB modulator. The LED D3 lights when the configuration phase is in progress and is off during normal operations. In system programming of the EEPROM allows to easily upgrade the modulator core and is accessible via the connector JP1, which accepts Atmel's standard programming cable and software.

At the connector JP2 are available the signals of the JTAG port of IC2. By means of this port it is possible to reprogram the FPGA with Xilinx software development toolkit and upload new core versions, overriding the EEPROM.

The 36 MHz crystal oscillator QG1 is the main clock source for the modulator core and the transmission DAC IC4.

The voltage regulator IC5 and the bipolar transistor Q1 provides the +3.3V and the +2.5V stabilized voltages required to drive IC1, IC2, IC3 and IC4. The modulator sinks 140 mA at pin 10 of JP3 where the stabilized +5Vdc power supply is applied. It is necessary that the external power supply is equipped with a large capacity electrolytic capacitor (at least 2,200 uF) or that, alternatively, it is able to provide a 500 mA peak current. This peak current is required by some milliseconds by the XC2S30 i.c. when it is powered up. The network formed by C18-R20 delays the internal +2.5V power supply for a short fraction of second allowing the charge of the external electrolytic capacitor which provides the extra current in the case that the external power supply peak current is less than 500 mA.

6. EXPERIMENTAL RESULTS

Fig. 11 shows a photo of the first sample of the digital SSB modulator Fig..12 and fig.13 shown the unwanted sideband rejection, measured with a 1 KHz input signal, and wide band spurious responses (1-17 MHz). The measures have been carried with a HP8563E spectrum analyzer. As shown by fig.12 the unwanted side band rejection is better than 80 dB. The quantization noise of the sigma delta AD converter is evident, it is confined in the bandwidth of the selected side band and its amplitude is quite low. In fig. 13 it is visible the highest spurious response generated by the modulator. Its amplitude is 75 dB lower than the p.e.p. output power. In the tests performed it has been verified also the spectral purity of a modulator in which the 14 bit DA converter AD9754 was substituted by a cheaper 10 bit DA converter (AD9760). The SFDR measured in this case was only slightly worse. The prototype with the 10 bit DAC showed just a greater noise floor, and still very low for amateur applications, in a bandwidth of a couple of megahertz around the centre frequency.

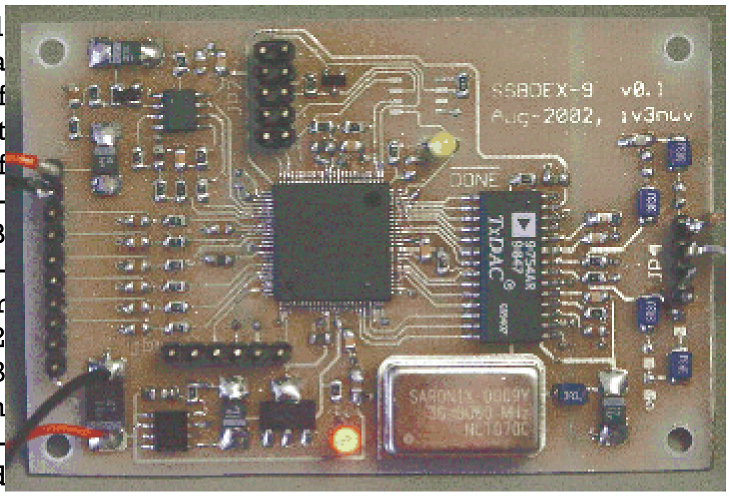

Fig.11 – A photo of the digital SSB modulator

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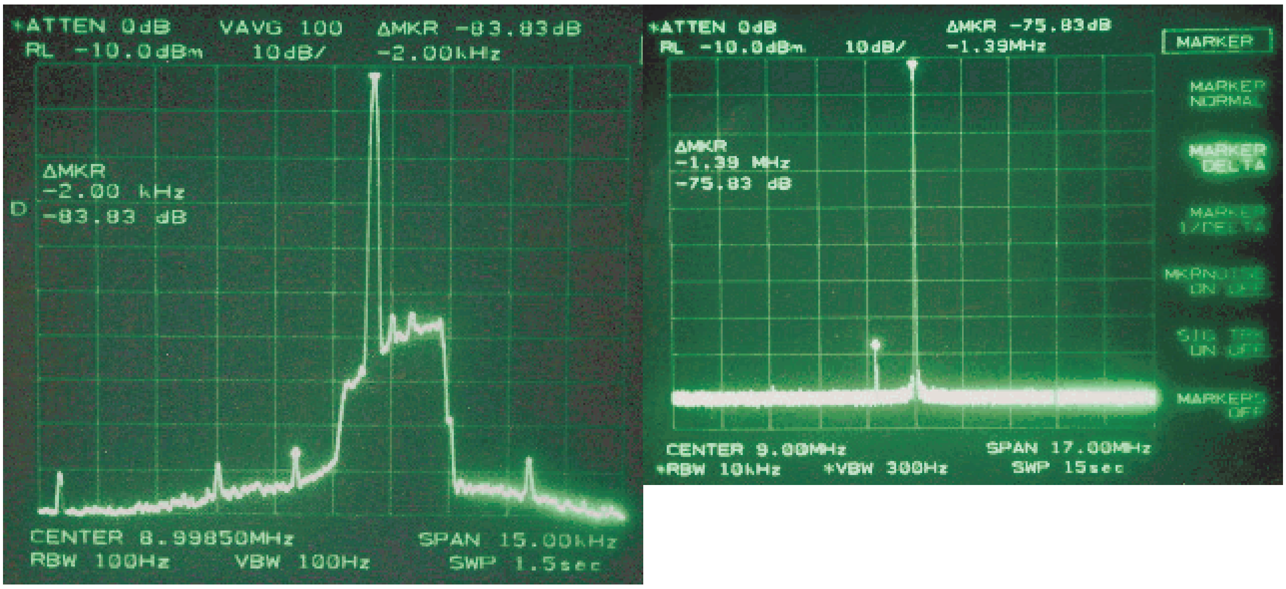

Fig.12,13 – Measures of the unwanted side band rejection and the wideband SFDR

7. CONCLUSIONS

Exploiting DSP algorithms optimised for the implementation on the hardware of arithmetic and trigonometric functions, a compact digital SSB modulator has been developed. The modulator can be used as the exciter of homebrew SSB/CW transmitters and the BFO of the receiver chain.

The logic core of the modulator was developed in VHDL language making large use of serial arithmetic modules. It fits into a cheap 30 Kgate FPGA (see fig. 14) and surpasses the performance of classic analogue SSB modulators. In future developments the digital SSB modulator and a CORDIC tuneable up converter will be integrated in a single gate array to realize a direct synthesis 0-30 MHz digital SSB/CW transmitter.

[Not Shown]
Fig. 14 – The core of the SSB modulator in a XC2S30 device

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Last update: 15 September 2002. Page accessed times.

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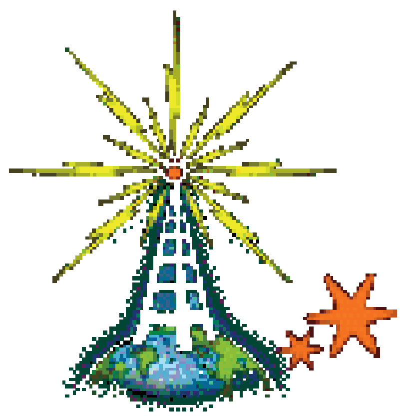

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