N94-22804

Direct Digital RF Synthesis and Modulation for MSAT Mobile Applications

Stewart Crozier, Ravi Datta, John Sydor
Department of Communications
Communications Research Centre
3701 Carling Avenue
P.O. Box 11490, Station "H"
Ottawa, Ontario, Canada K2H 8S2
Telephone: (613)998-2388
Facsimile: (613)990-6339

ABSTRACT

A practical method of performing direct digital RF synthesis using the Hilbert transform single sideband (SSB) technique is described. It is also shown that amplitude and phase modulation can be achieved directly at L-band with frequency stability and spurii performance exceeding stringent MSAT system requirements.

INTRODUCTION

Direct synthesis of RF modulation has always been a touchstone concept with radio terminal designers. It has always held the promise of hardware reduction with the achievant of superior performance. However, most direct synthesis achievements have been at frequencies substantially lower than those that will be used with planned mobile satellite and personal communications systems. For the current MSAT and INMARSAT mobile satellite systems, specifications for long term frequency stability, phase noise, spurious emissions, and channel control significantly exceed the capabilities of existing direct digital synthesizers, which are limited primarily by the speed and resolution of their digital to analog converters (DACs). In view of the current pace of DDS technology, even when the limitations of the DACs are overcome, achievement of MSAT performance objectives at L-Band by DDS technology will not materialize until after the launch and commissioning of MSAT. When such technology does materialize there is no guarantee that it will be available at a low enough cost to make its incorporation into terminals viable. Up to now conventional Phase-Lock loop technologies or combined Phase-locked loop/DDS hybrid technologies seem the most promising frequency synthesis schemes for narrow band L-Band signal processing.

In view of the current limitations of DDS technology, CRC along with private researchers [1] has developed a new RF digital modulation technique that does not rely on the extensive use of phase lock loops and mixer/filter upconversion stages. The technique is extremely versatile and can be implemented at low cost. Phase and amplitude modulation can be achieved directly at L-Band with frequency stability and spurii performance exceeding stringent mobile satellite system requirements. This new technique is based on a unique hybrid of technologies, all of which are mature and highly amenable to low cost mass production practices. As a consequence this technique holds the promise of being able to bypass the problems limiting the evolution of DDS as it is currently envisioned.

HILBERT TRANSFORM SSB GENERATION

Any modulating signal f(t) can be upconverted to a higher frequency by using the Hilbert transform (phase shifted SSB) technique [2]. In order to do this both the modulating signal and the carrier frequency must be generated as quadrature components and mixed by balanced modulators (Fig.1). Each modulator outputs a double sideband signal, $f_{d1}(t)$ and $f_{d2}(t)$.

$$f_{d1}(t) = A \cos\omega_m t \cos\omega_c t$$

$$= \frac{A}{2} [\cos(\omega_c + \omega_m)t + \cos(\omega_c - \omega_m)t] \quad (1)$$

$$f_{d2}(t) = A \sin\omega_m t \sin\omega_c t$$

$$= \frac{A}{2} [\cos(\omega_c - \omega_m)t - \cos(\omega_m + \omega_c)t] \quad (2)$$

By subtracting or adding the outputs of each modulator, we generate either the upper or lower sideband, i.e.:

$$F_o(t) = A \cos(\omega_m + \omega_c)t \quad (3)$$

In (3) we see that our modulating signal is shifted up by the carrier frequency. By exercising coarse control at a high frequency with the carrier, and a fine control at the lower modulating frequency, it is possible to exercise a wide band, fine control at an offset from the carrier frequency.

One major problem with the design of Hilbert Transform SSB modulators is the preservation of the $\pi/2$ phase shift over a wide band of frequencies. Failure to do so results in the emergence of the cancelled sideband. A phase error of 1 degree will result in the cancellation of the undesired sideband by only 35 dB. Most mobile satellite specifications require sprurii emissions to be in excess of 65 dBC. A second serious problem that exists with the practical implementation of the Hilbert transformation is the leakage of the carrier through the balanced modulators. Low cost suppressed carrier balanced modulators in practice cannot reject the carrier by more than 25 dB. This is why in most practical SSB radios the modulated output is filtered by a highly selective filter prior to further up-conversion.

To overcome these inherent problems, yet take advantage of the frequency control capability demonstrated in (3), requires modification to the basic Hilbert transform circuit shown in Figure 1.

FEEDBACK CONTROLLED HILBERT TRANSFORM SYNTHESIZER

Figure 2 shows one approach that can be taken to mitigate the hardware limitations of the Hilbert transform circuit. What is shown is an SSB RF synthesizer that can produce outputs at L-Band which can be stepped in 1 Hz increments and have phase noise lower than -80 dBC at 1Khz from the carrier. Spurii content lower than -60 dBC is also achievable.

The circuit has a low frequency dual DDS synthesizer that generates quadrature outputs. For example, this circuit and its DACs may produce a signal from 0.1 to 1.1 MHz in 1 Hz steps. The DDS is driven by a 20 MHz crystal which is divided down within the DDS, hence the phase noise of the output of the DDS is that of its crystal reference (F_{ref}) improved by 20 log (f_{ref}/f_{output}) dB. However this improvement is limited by the noise floor of the DDS circuitry (-150 dBC). What limits the spectral purity of the device is the performance of the 12 bit digital to analog converters which have spurious deficiencies due to glitches, output settling, and nonlinearity; with glitches being the main contributor. Theoretically the digitized sine output of a 12 bit DAC will provide a broadband signal-to-spurious ratio of 72 dB.

The output of the quadrature DDS is sent to the Hilbert transform circuit and to the balance circuit, which will be discussed later. A second signal is injected into the Hilbert transformer circuit from a coarse step synthesizer. This can be as high as 1600 MHz, however, with our experimental circuits we use a phase lock loop synthesizer operating in the 50-400 MHz range. This synthesizer has 1 MHz step sizes and is a single loop device with a very wide loop filter (10-30 Khz wide). Consequently the microphonic and spurii problems are significantly mitigated with this synthesizer. Phase noise for this device is of the order of -80 dBC at 1 kHz offset.

The combination of the DDS and the single loop PLL set the performance limits on the output RF signal. It is the last circuit, the balance circuit, which eliminates the most deleterious sources of signal degradation with the Hilbert transformer SSB synthesizer. Details regarding this circuit are covered in [1]. Within the balance circuit error signals are generated which are proportional to the carrier leakthrough and the phase and amplitude errors causing the undesired sideband signal.

Carrier leakthrough is determined by measuring the level of the DC signal that is generated by mixing the RF output of the modulator with the carrier signal generated by the PLL. If the RF output of the modulator contains the carrier signal then this will be detected by the mixing process. The resulting error signal is used to drive DC biasing circuits which change the DC offset bias on the balanced modulators within the Hilbert transform modulator circuit. By adjusting the DC bias it is possible to eliminate carrier feedthrough with the balanced modulators.

Suppressing the undesired sideband signal is a somewhat more complex process, but in essence involves further mixing of the DDS signals and RF output. The balance circuit produces a phase error control signal which is used to adjust the phase of the I and Q signals from the DDS. An amplitude error control signal is also generated to control the amplitude of one of the DDS signals.

Because of its feedback nature this hybrid synthesizer has a time delay on the order of 10 milliseconds between the time a frequency instruction is loaded into it and the time the required RF signal is synthesized. Figures 3 and 4 show the typical RF performance characteristics of the synthesizer

The ability to control the frequency and phase of the output signal to such a high degree of accuracy makes it possible to use the synthesizer as a signal modulator.

DIRECT MODULATION APPROACH

Figure 5 shows the block diagram of a direct RF modulator. The system consists of a microprocessor with FM and AM look-up tables, a feedback controlled Hilbert transform synthesizer, as described earlier, and an amplitude modulator.

The approach taken to generate an arbitrary modulation format is to take the desired complex baseband modulation signal and break it up into equivalent FM and AM signals. The FM signal is then generated by the DDS and translated directly to the RF frequency using the Hilbert transform SSB technique. The AM portion is applied to the FM signal directly at RF using a mixer.

Consider the complex baseband MPSK transmit signal given by

$$s(t) = \sum_k a_k\, h(t-kT) \tag{4}$$

where

$$a_k = \exp(j2\pi m/M), \quad m=1..M \tag{5}$$

represents the complex data symbols, M is the number of phases, and h(t) is the impulse response of the desired baseband transmit filter. For example, QPSK modulation with M=4 and a 60% roll-off root-raised-cosine (RRC) filter is specified for the MSAT communications channels. The equivalent FM and AM signals are given by

$$s_{FM}(t) = \frac{d}{dt}\, \text{phase}[s(t)] \tag{6}$$

$$s_{AM}(t) = |s(t)| \tag{7}$$

The bandwidth of the FM and AM signals is much wider than the bandwidth of the desired filter response, h(t). This implies that the sampled versions of the FM and AM signals must have many samples per symbol period to ensure that a sufficient number of FM and AM sidelobes are properly characterized and that the undesired sampling images are suppressed to acceptable levels. It is difficult to compute the FM and AM signals in real time because of the high sampling rates involved and the calculations required to convert from cartesian to polar coordinates. A table lookup approach was used for this reason. It was found that 32 samples per symbol period is sufficient to force the FM images down more than 70 dB with no additional band-pass filtering required at the output of the DDS. This allows the DDS to be used for both FM modulation and channelization over several MHz of bandwidth. Using 32 samples per symbol period for the AM portion as well resulted in excellent performance. The AM sampling images fall off much slower than the FM images, but the AM images are easily suppressed at baseband using a simple low-pass filter at the output of the AM DAC.

The table lookup approach stores one symbol period of the FM and AM waveforms for all possible bit patterns of a given length. The table size is an exponential function of the length, L, of the desired impulse response, h(t). With differential encoding encorporated into the FM and AM tables, the FM and AM waveforms are independent of the phase of the first symbol spanned by h(t). This makes both tables M times smaller than one might initially expect, and differential encoding is free. The required table size in bits is given by

$$\text{table size} = B\ N\ M^{L-1} \quad \text{bits} \qquad (8)$$

where B is the number of bits per sample and N is the number of samples per symbol. As an example, a good approximation for the 60% RRC filter has been found which easily meets the MSAT specs using a span of only L=5 symbol periods. With B=16, N=32, and M=4 for QPSK, the FM and AM tables are only 128 K bits each.

Figure 6 shows the reconstructed spectrum obtained with these parameters. The first sidelobe, at a frequency of 1 symbol rate, is due to the filter h(t) and is down 45 dB. The first image, at a frequency of 32 symbol rates, is mostly due to the unfiltered FM spectrum and is down about 75 dB.

CONCLUSIONS

A practical method of performing direct digital RF synthesis using the Hilbert transform SSB technique has been described. It was also shown that amplitude and phase modulation can be achieved directly at L-band with frequency stability and spurii performance exceeding stringent MSAT system requirements. Related research being conducted at CRC indicates that a direct RF down-converter/demodulator is also feasible.

References

[1] U.S. Patent 5162763; K. Morris, *Single Sideband Modulator for Translating Baseband Signals to Radio Frequency in single stage*, November 10, 1992.

[2] Single Sideband Issue, Proceedings of the I.R.E. Vol. 44, No. 12, Dec 1956, .

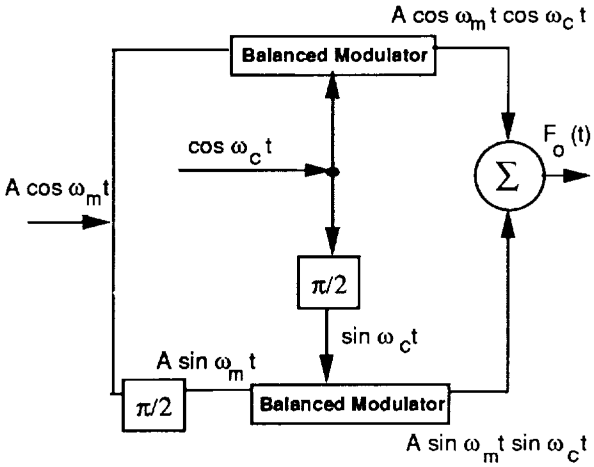

Figure 1 Hilbert Transform SSB Circuit

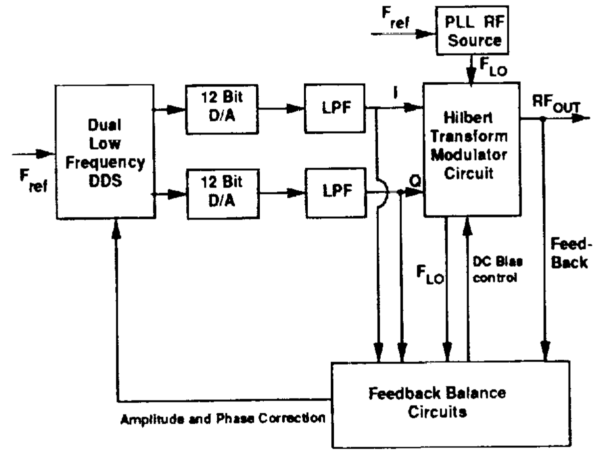

Figure 2
Hybrid DDS/PLL Hilbert Transform Synthesizer

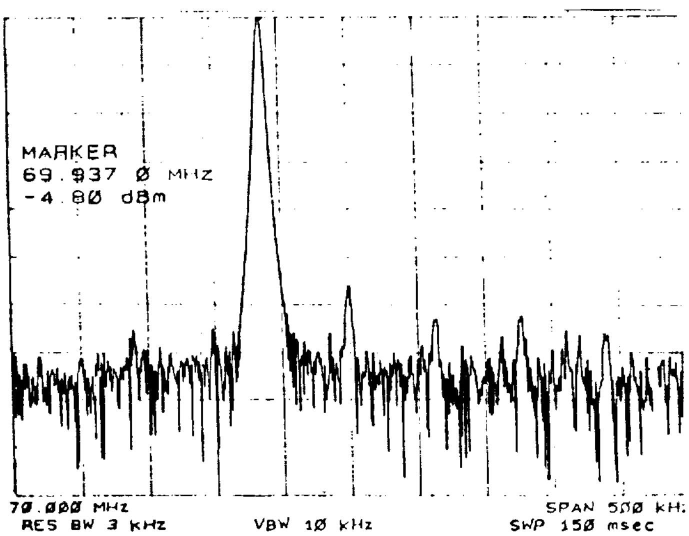

Figure 3
Synthesizer Output - Wide Band Width

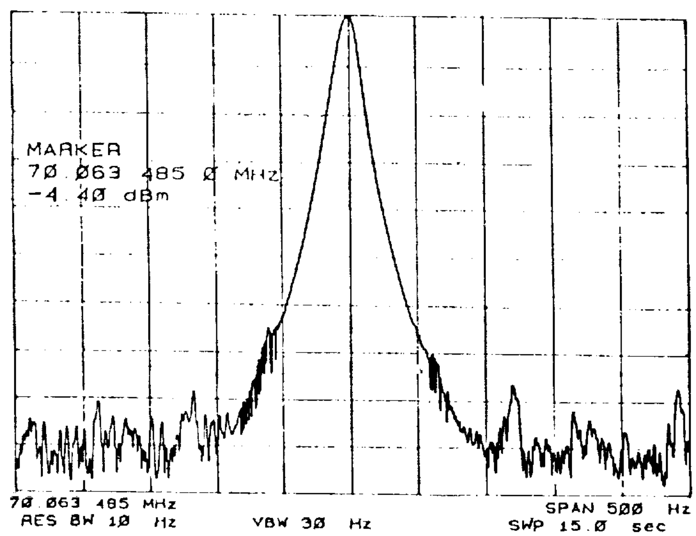

Figure 4 Synthsizer Output - Narrow Band Width

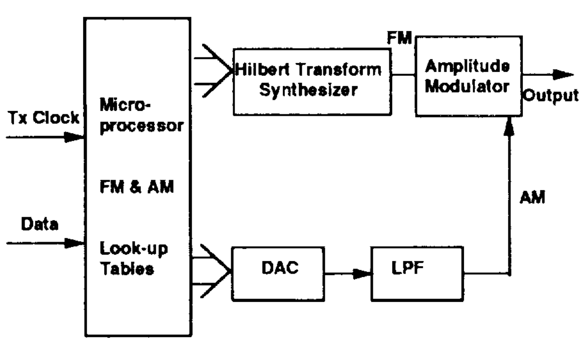

Figure 5
Block Diagram of Direct RF Modulator

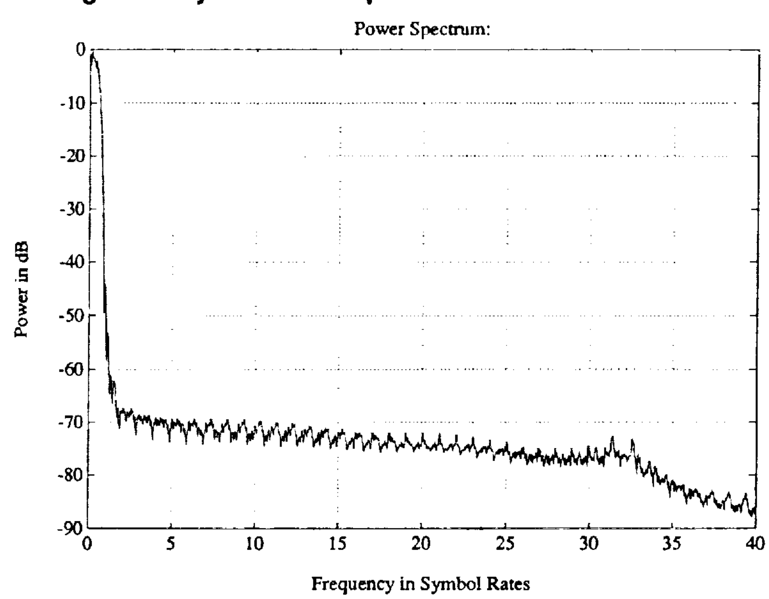

Figure 6
Power Spectrum of Reconstructured Signal