Implementing Modulation Functions in Microwave Frequency Synthesizers

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This article reviews the typical methods for implementing modulation within a frequency synthesizer instrument or system

The main function of a frequency synthesizer is to deliver a stable and clean signal. However, many applications require not only a fixed frequency signal but also various modulation functions ranging from simple pulse, amplitude, frequency and phase modulation to complex digital modulation formats. Although modulators are usually realized as external devices, they can also be incorporated into a frequency synthesizer core. Inside any synthesizer there are many circuits that can carry multiple functions and be reused to increase the functionality without a significant increase in cost. This results in a more cost efficient and versatile design. The most commonly used modulation schemes are briefly reviewed in this article. Further details on modulation theory and implementation techniques can be found in [1-7].

Pulse Modulation

Pulse modulation is probably the simplest modulation form. It is achieved by switching the output signal on and off in accordance with the applied modulating pulses. The result is a sequence of RF pulses that replicate (or tend to replicate) the input modulating signal. The minimum RF pulse width, rise time, fall time and overshoot are important characteristics that define how well the modulating signal is replicated. Typical rise time and fall time numbers required are in the order of 10 nanoseconds. Pulse modulation on/off ratio is another critical parameter. A typical specification is 80 dB or higher. The modulating signal frequency (also called *rate*) can be between DC and several megahertz. Pulse modulation is practically implemented by inserting a switch (or a chain of switches for a higher on/off ratio) into the synthesizer output path as depicted in Figure 1. The switch can be built using PIN-

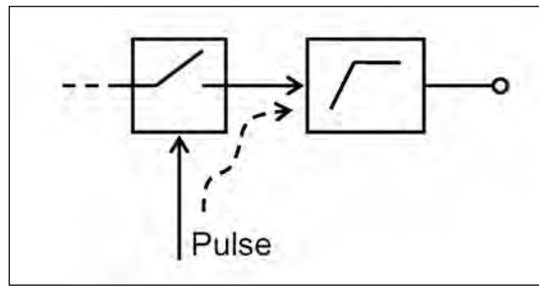

Figure 1 · Pulse modulation is implemented by inserting a switch into the synthesizer output path.

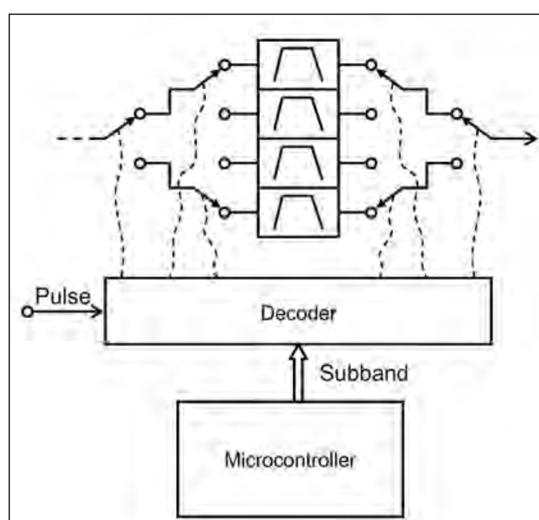

Figure 2 · A pulse modulator is integrated into a switched filter bank.

diodes or FET devices that support nanosecond switching. A high-pass filter follows the switch to suppress the leakage of the modulating signal (called *video feedthrough*) to the synthesizer output.

Alternatively, the pulse modulator can be conveniently combined with a switched filter bank used for harmonic (or subharmonic) rejection. The idea is to reduce the design complexity and cost by utilizing the same devices for both functions. Furthermore, no additional loss is introduced that eases requirements for the output power amplifier. In this case, the pulse modulator incorporates a digital decoder (as shown in Fig. 2) to control the switches in such a manner that they will provide the highest possible isolation (on/off ratio) for any given frequency subband.

Amplitude Modulation

Amplitude modulation (AM) historically has been one of the most popular methods for carrying information via RF frequencies. It is realized by varying the output signal amplitude in accordance with an applied modulating signal. The simplest way to implement AM is to control the insertion loss of an attenuator inserted into the synthesizer output circuit as depicted in Figure 3. This can be naturally combined with an open-loop amplitude control as depicted in Figure 4. The synthesizer is first commanded to set a desired output power level by programming DAC (digital-to-analog converter) voltage. Then a modulating voltage is applied over the DAC voltage to vary the output signal around its nominal value. Naturally, the output power cannot be set at its highest (or lowest) level because certain headroom is needed to allow further power changes. The maximum power variation (which can also be expressed in terms of modulation index or depth) is achieved by setting the output power level in the middle of its control range. Another important requirement is linearity because the modulator must translate the modulating signal with minimal distortion. This may further limit a realizable modulation depth. Various linearization techniques can be applied to minimize AM signal distortion. In some cases, it is desirable to implement not a linear but a logarithmic modulating scale, meaning that the output power changes in dB per volt. This mode is utilized for large power variations (e.g., for simulation of rotating antenna patterns) and is called deep AM.

Alternatively, amplitude modulation can be implemented by summing the modulating signal into the ALC (automatic level control) loop as shown in Figure 5. In general, the ALC-based amplitude modulation offers better linearity and repeatability characteristics. However, the modulation depth may be limited by the available ALC dynamic range, which, in turn, depends on the utilized detector. The maximum modulating signal rate is also lower compared to the open-loop alternative because of the settling time of the closed-loop ALC system.

Frequency and Phase Modulation

Frequency modulation (FM) is another popular form of analog modulation that offers better signal immunity compared to AM. The process of producing a frequency-modulated signal involves the variation of the synthesizer output frequency in accordance with the modulating signal. The frequency bandwidth where the synthesized signal fluctuates is proportional to the peak amplitude of the modulating signal and is called *frequency deviation.* FM

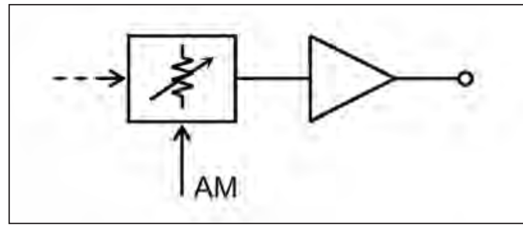

Figure 3 · Amplitude modulation is realized using a voltage-controlled attenuator.

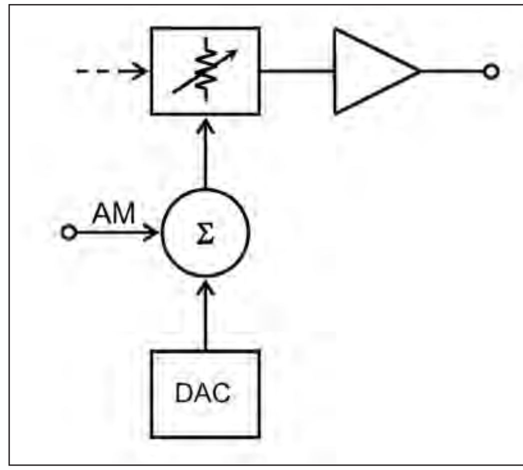

Figure 4 · Amplitude modulation is combined with an open-loop amplitude control.

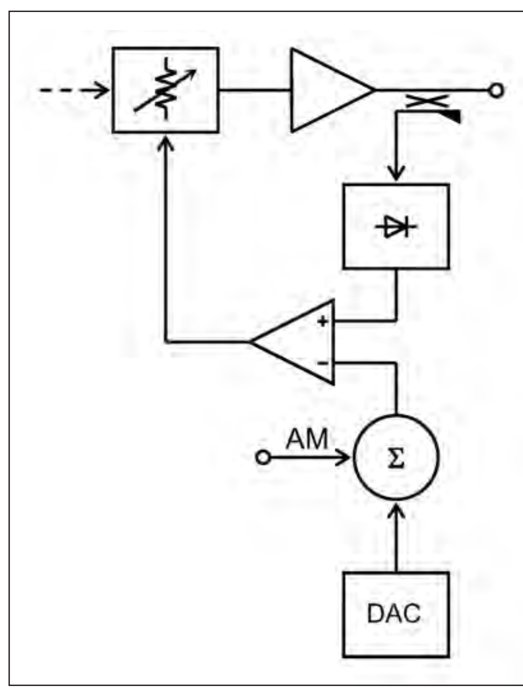

Figure 5 · Amplitude modulation can be realized by summing the modulating signal into the ALC loop.

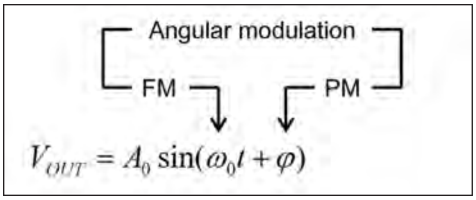

Figure 6 · Frequency and phase modulation.

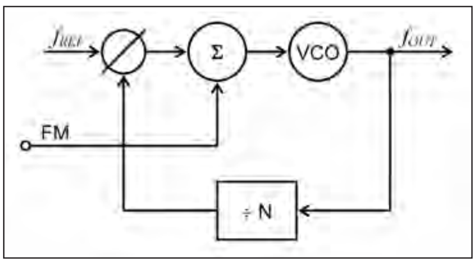

Figure 7 · Frequency modulation is realized by modulating the VCO tuning voltage.

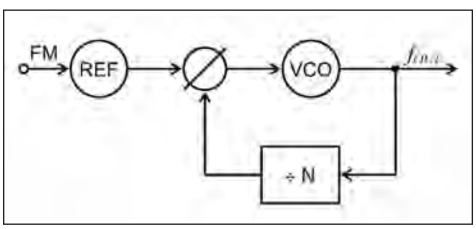

Figure 8 · A modulating signal is applied to the reference oscillator.

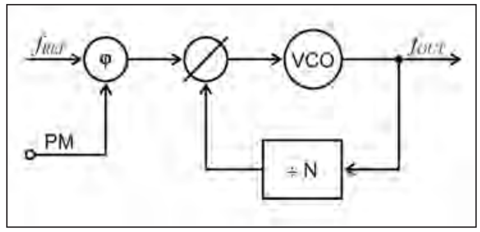

Figure 9 · A phase shifter provides a phase modulation function.

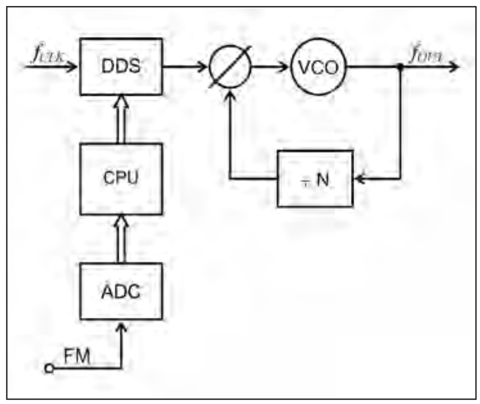

Figure 10 · Frequency modulation is realized by controlling the DDS output frequency.

can also be described by *modulation index,* which is the ratio of the maximum frequency deviation to the frequency of the modulating signal.

Note that we can vary not only the frequency but also the phase of the synthesized signal, thus producing phase modulation (PM). Both processes are quite similar because in both cases we vary the argument (the angle) of the same sine function as illustrated in Figure 6. Hence, the angular modulation is a more general case that represents both FM and PM. The difference is not in the output signal waveform but rather in the modulator circuit configuration; that is, what parameter (frequency or phase) is directly proportional to the amplitude of the modulating signal. Because the instantaneous angular frequency is mathematically the time derivative of the phase, it is possible to convert FM to PM (and vise versa) by adding an integrator (or differentiator) circuit into the modulating signal path.

How can we modulate the synthesizer output frequency? From first glance, it is quite straightforward—we can simply modulate (i.e., change) the VCO (voltage-controlled oscillator) tuning voltage around the value where it is settled. The problem, however, is that the synthesizer's PLL (phase-lock-loop) core will tend to correct any voltage change we introduce. Most likely, we will lose this battle unless we change the tuning voltage so fast that the PLL will not be able to react to the change. This is exactly the idea that stands behind a so-called wideband FM modulation mode. The FM modulator is built by adding a circuit (e.g., an operational amplifier) that sums an external modulating signal with the control voltage delivered by PLL as depicted in Figure 7. The PLL remains locked all the time, thus ensuring that the overage output frequency remains correct. For proper operation, the modulating signal rate has to be well above the loop filter bandwidth. Thus, the PLL filter bandwidth is adjusted (narrowed down) to allow lower modulating rates. As a result, the phase noise usually increases when FM is enabled. Typical achievable modulating rates range from a few kilohertz to tens of megahertz.

What if we need to apply a lower-frequency modulating signal? Obviously, we have to further decrease the loop filter bandwidth, which may not always be possible because of prohibitory high VCO free-running phase noise at low frequency offsets. An alternative solution is to modulate not the VCO but the reference oscillator as shown in Figure 8. If the modulating signal rate is sufficiently low, the PLL will track the reference frequency change and, hence, translate the modulation to the VCO output. This mode is often called narrowband FM because the modulating frequency must be within PLL filter bandwidth. The loop filter bandwidth should be set as wide as possible to allow higher modulating rates. Typical rates start from nearly DC to a few tens of kilohertz. Thus, the narrowband mode complements its wideband counterpart to extend the overall modulating frequency range.

A disadvantage of this technique is low achievable deviation caused by very low tuning sensitivity of the reference oscillator. Although the reference frequency deviation is multiplied up by the PLL at the 20log*N* rate, the synthesizer output deviation may be insufficient. A higher deviation can be achieved by changing not the reference frequency but rather its phase as depicted in Figure 9. This represents a classical phase modulation; however, both forms are interchangeable. Practical implementation requires a phase shifter that can be purchased

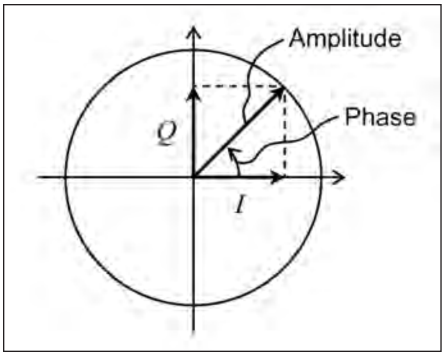

Figure 11 · A signal is presented as a vector on a polar diagram.

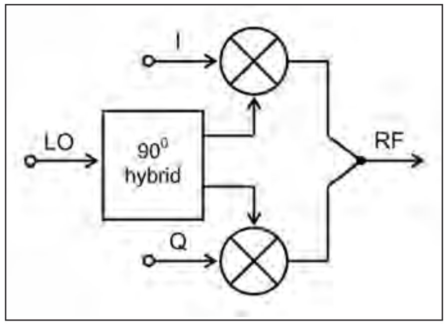

Figure 12 · IQ-modulator block diagram.

or can be built using discrete devices such as varactor diodes.

An interesting solution is to control the division ratio of a frequency divider inserted into either the PLL reference or feedback path as illustrated in Figure 10. The divider has to be a high-resolution device such as a fractional-*N* divider or DDS (direct digital synthesizer). The modulating signal is first digitized by an ADC (analog-to-digital converter) and then is summed with the DDS tuning word to vary the DDS's output frequency. Because the DDS offers exceptionally small frequency increments and a fast update rate, a simple yet high-performance FM (or PM) modulator can be constructed.

Complex Modulation

More effective modulation formats are possible by simultaneously varying both amplitude and phase. The simplest way to visualize such a complex signal is to draw it as a vector on a polar diagram. The amplitude and phase are represented as the length and the angle of the vector as shown in Figure 11. In digital communication systems, such a signal is expressed in I (in-phase) and Q (quadrature) terms, which are projections of the signal vector on a corresponding orthogonal axis. Therefore, the amplitude and phase modulation assumes the change of the signal vector, which can be conveniently accomplished by varying two independent IQ-components. Hence, such a complex modulation is called *vector* or *IQ-modulation*.

Vector modulation can be applied directly at RF frequencies by utilizing an IQ-modulator. It consists of two identical mixers driven with a 90-degree phase shift at their LO (local oscillator) ports as shown in Figure 12. The base-band data signals are applied directly to the mixer IF (intermediate frequency) ports, upconverted, and summed together with no phase shift between them. The resulting output is an IQ-modulated signal at the same carrier frequency as the LO. The quality of the synthesized signal can be tested by applying two base-band signals of the same frequency and amplitude with a 90-degree phase shift with respect to each other. For a perfect modulator, only one sideband should be present. However, in reality, the output signal contains another sideband because of imperfect amplitude and phase balance. For example, equalizing the signal paths within 1 dB (amplitude) and 10 degrees (phase) results in approximately a 20-dB sideband rejection. Naturally, better rejection is required. Moreover, an LO leakage also takes place. The undesired sideband can be further suppressed by adjusting the amplitude and phase of the applied IQ signals. The LO leakage can be controlled by adjusting DC offset voltages for the diodes used in the balanced mixers. Therefore, it is generally possible to calibrate the modulator characteristics to a degree where it can be practically utilized. The difficulty is that this calibration has to be implemented at many frequencies across the entire operating range. Moreover, the calibration has to survive over time and temperature changes. Thus, achieving a good image and LO leakage suppression for a broadband, high-frequency, direct IQ-modulator is a very challenging task.

An alternative solution is to create a desired IQ-modulated signal at a lower, fixed frequency and then upconvert it to microwave frequencies as illustrated in Figure 13. Obviously, it is much easier to achieve better cancellation of undesired products at a single (and lower frequency) point. However, the difficulty now moves to the upconversion side. We still need to remove the undesired sideband and LO leakage posted by the second (regular) mixer. However, because the product separation is much larger (compared to the direct IQ-modulation), a hardware filter can be used. For a broadband operation, a YIG-tuned filter is a simple and effective solution. The disadvantage of the YIG filter is slow tuning speed and relatively narrow pass-band that can be insufficient in certain applications. A switched filter bank (Fig. 14) offers better characteristics. However, it requires a larger number of channels (compared to devices used for harmonic and sub-harmonic filtering) and, hence, is hardware extensive. This results in a more complex system design and posts other challenges (e.g., achieving high isolation between filter channels, etc.).

In the past, complex microwave assemblies were often built using individual connectorized modules connected with coaxial cables. The designer could easily isolate and refine individual blocks to make them perfect. These days, such assemblies have to be made on a common PCB using tiny surface-mount parts. A great effort is required to minimize interactions between individual components sitting on the same board. Furthermore, many parts are reused

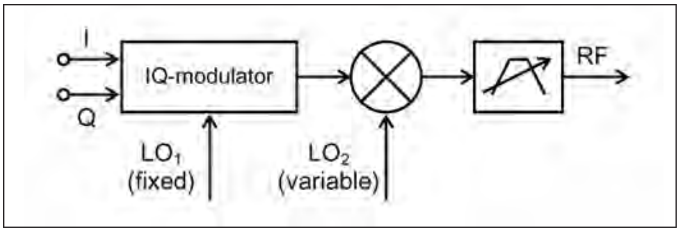

Figure 13 · Upconversion using a tunable filter.

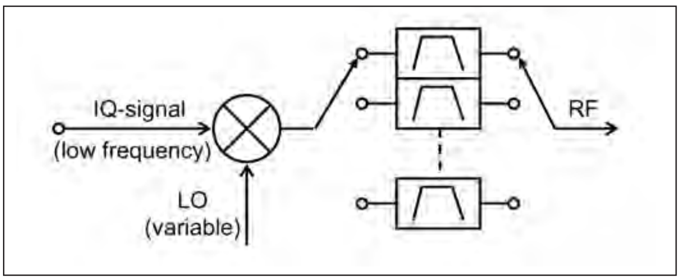

Figure 14 · A broadband upconverter based on a switched filter bank.

to accomplish different functions, which are distributed through the whole assembly. The net result is a significant increase in "design density," meaning both component count and functionality per square inch. All these factors drastically complicate the design process. Nevertheless, this seems to be a "must" approach these days.

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