

Agilent 33220A
Function/Arbitrary Waveform Generator

TUTORIAL

pages from User's Guide

Tutorial

Tutorial

In order to achieve the best performance from the Agilent 33220A, it may be helpful for you to gain a better understanding of the internal operations of the instrument. This chapter describes basic signal-generation concepts and gives specific details on the internal operations of the function generator.

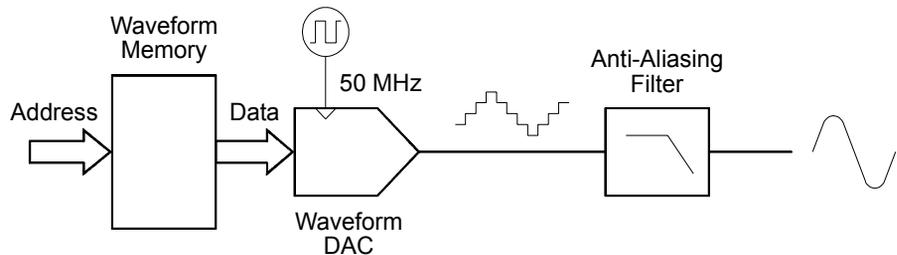
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You can use an arbitrary waveform generator in a variety of applications where it might be otherwise difficult or impossible to generate complex output waveforms. With an arbitrary waveform generator, signal imperfections such as rise time, ringing, glitches, noise, and random timing variations can be easily simulated in a controlled manner.

Physics, chemistry, biomedicine, electronics, mechanics, and other fields can benefit from the versatility of an arbitrary waveform generator. Wherever things vibrate, pump, pulse, bubble, burst, or change with time in any way, there are possible applications – limited only by your ability to specify the waveform data.

Direct Digital Synthesis

The Agilent 33220A uses a signal-generation technique called *Direct Digital Synthesis* (DDS) for all waveform functions except pulse. As shown below, a stream of digital data representing the desired waveform is sequentially read from waveform memory and is applied to the input of a digital-to-analog converter (DAC). The DAC is clocked at the function generator's sampling frequency of 50 MHz and outputs a series of voltage steps approximating the desired waveform. A low-pass "anti-aliasing" filter then smooths the voltage steps to create the final waveform.

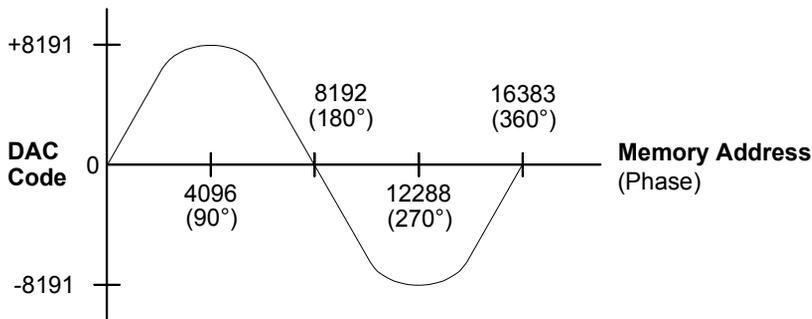


Direct Digital Synthesis Circuitry

The 33220A uses two anti-aliasing filters. An elliptical filter is used for continuous sine waves because of its nearly flat passband and sharp cutoff above 20 MHz. Since elliptical filters exhibit severe ringing for waveforms other than continuous sine waves, a linear-phase filter is used for all other waveform functions.

For standard waveforms, and arbitrary waveforms that are defined with fewer than 16,384 (16K) points, the function generator uses waveform memory that is 16K words deep. For arbitrary waveforms that are defined with more than 16K points, the function generator uses waveform memory that is 65,536 (64K) words deep.

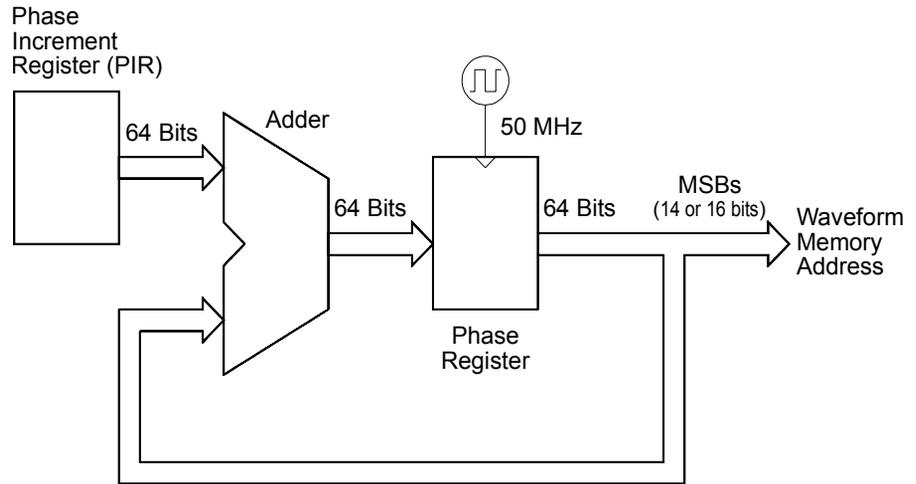
The 33220A represents amplitude values by 16,384 discrete voltage levels (or 14-bit vertical resolution). The specified waveform data is divided into samples such that one waveform cycle exactly fills waveform memory (see the illustration below for a sine wave). If you create an arbitrary waveform that does not contain exactly 16K or 64K points, the waveform is automatically “stretched” by repeating points or by interpolating between existing points as needed to fill waveform memory. Since all of waveform memory is filled with one waveform cycle, each memory location corresponds to a phase angle of $2\pi/16,384$ radians or $2\pi/65,536$ radians.



Sine Wave Representation in Waveform Memory

Direct digital synthesis (DDS) generators use a *phase accumulation* technique to control waveform memory addressing. Instead of using a counter to generate sequential memory addresses, an “adder” is used (see the following page). On each clock cycle, the constant loaded into the phase increment register (PIR) is added to the present result in the phase accumulator. The most-significant bits of the phase accumulator output are used to address waveform memory. By changing the PIR constant, the number of clock cycles required to step through the entire waveform memory changes, thus changing the output frequency.

The PIR determines how fast the phase value changes with time and thereby controls the frequency being synthesized. More bits in the phase accumulator result in finer frequency resolution. Since the PIR affects only the rate of change of the phase value (and not the phase itself), changes in waveform frequency are phase-continuous.



Phase Accumulator Circuitry

The 33220A uses a 64-bit phase accumulator which yields $2^{-64} \times 50$ MHz or 2.7 picohertz frequency resolution internally. Note that only the 14 or 16 most-significant bits of the Phase Register are used to address waveform memory. Therefore, when synthesizing low frequencies (less than 3.05 kHz for a typical, 16K point waveform), the address will not change in every clock cycle. However, at higher frequencies (greater than 3.05 kHz), the address will change by more than one location during each clock cycle and some points will be skipped. If too many points are skipped, a phenomenon known as “aliasing” will occur and the waveform output will become somewhat distorted.

The *Nyquist Sampling Theorem* states that in order to prevent aliasing, the highest frequency component of the desired output waveform must be **less than half** of the sampling frequency (25 MHz for the 33220A).

Creating Arbitrary Waveforms

The Agilent 33220A allows you to create arbitrary waveforms of up to 64 K points (65,536 points), and provides five built-in examples of arbitrary waveforms. You can create an arbitrary waveform from the front panel, or you can use the Agilent IntuiLink software provided on the CD-ROM included with the Agilent 33220A. The Agilent IntuiLink software allows you to create arbitrary waveforms using a graphical user interface on your PC, and then download them into the Agilent 33220A. You can also capture waveforms from your Agilent oscilloscope and import them into IntuiLink. Please see the online help included with the Agilent IntuiLink software for further information.

For most applications, it is not necessary to create an arbitrary waveform with a specific number of points since the function generator will repeat points (or interpolate) as necessary to fill waveform memory. For example, if you specify 100 points, each waveform point will be repeated an average of $16,384 / 100$ or 163.84 times. For the 33220A, you do not have to change the length of the waveform to change its output frequency. All you have to do is create a waveform of any length and then adjust the function generator's output frequency. However, in order to get the best results (and minimize voltage quantization errors), it is recommended that you use the full range of the waveform DAC.

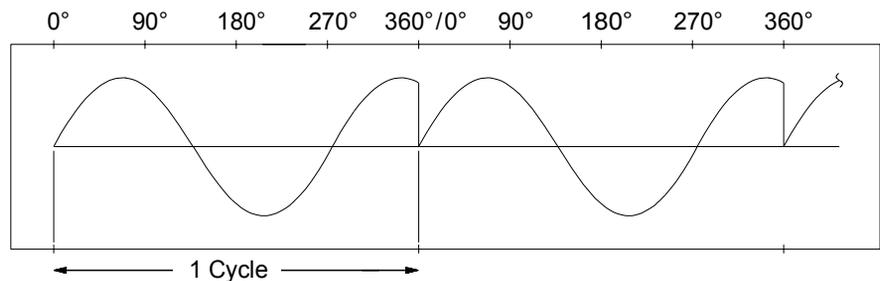
When entering waveform points from the function generator's front panel, you are not required to enter the points at evenly-spaced intervals in time. You can always add additional points as needed where the waveform is more complex. From the front panel only, you can also use linear interpolation to smooth the transition between waveform points. These features make it possible to create useful arbitrary waveforms using a relatively small number of points.

With the 33220A, you can output an arbitrary waveform to an upper frequency limit of 6 MHz. However, note that the *useful* upper limit is usually less due to the function generator's bandwidth limitation and aliasing. Waveform components above the function generator's -3 dB bandwidth will be attenuated.

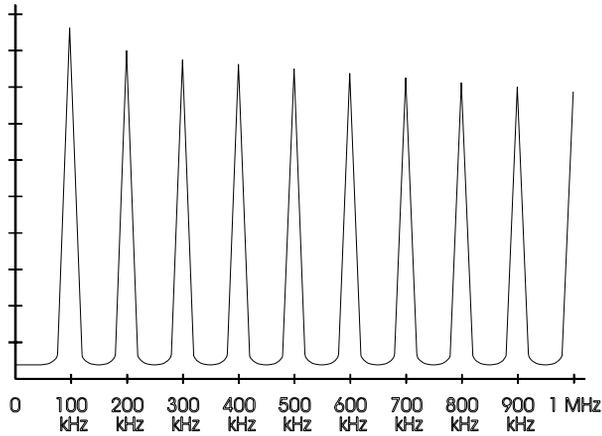
For example, consider an arbitrary waveform consisting of 10 cycles of a sine waveform. When you set the function generator's frequency to 1 MHz, the actual output frequency will be 10 MHz and the amplitude will be attenuated by approximately 3 dB. As you increase the frequency above 1 MHz, more attenuation will occur. At approximately 2.5 MHz, waveform distortion due to aliasing will become evident. Some aliasing will be present in most arbitrary waveforms, but whether or not it will be troublesome depends on your specific application.

When creating arbitrary waveforms, the function generator will always attempt to replicate the finite-length time record to produce a periodic version of the data in waveform memory. However, as shown below, it is possible that the shape and phase of a signal may be such that a discontinuity is introduced at the end point. When the waveshape is repeated for all time, this end-point discontinuity will introduce *leakage errors* in the frequency domain because many spectral terms are required to describe the discontinuity.

Leakage error is caused when the waveform record does not include an integral number of cycles of the fundamental frequency. Power from the fundamental frequency, and its harmonics, is transferred to the spectral components of the rectangular sampling function. You can reduce leakage errors by adjusting the window length to include an integer number of cycles or by including more cycles within the window to reduce the residual end-point discontinuity size. Some signals are composed of discrete, non-harmonically related frequencies. Since these signals are non-repetitive, all frequency components cannot be harmonically related to the window length. You should be careful in these situations to minimize end-point discontinuities and spectral leakage.



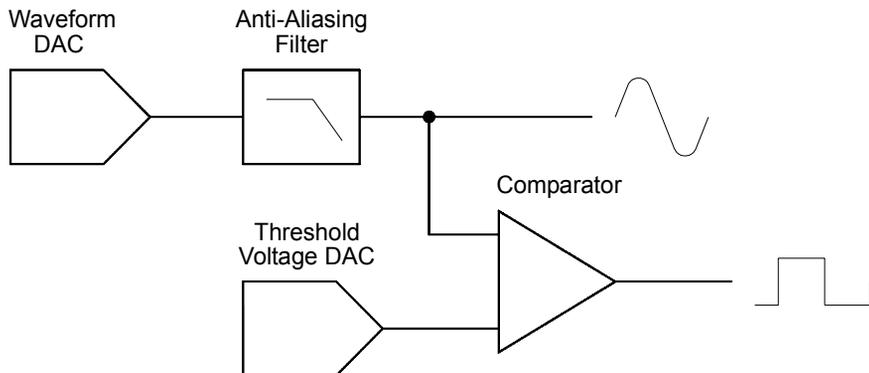
Arbitrary Waveform with Discontinuity



Spectrum of Above Waveform at 100 kHz

Square Waveform Generation

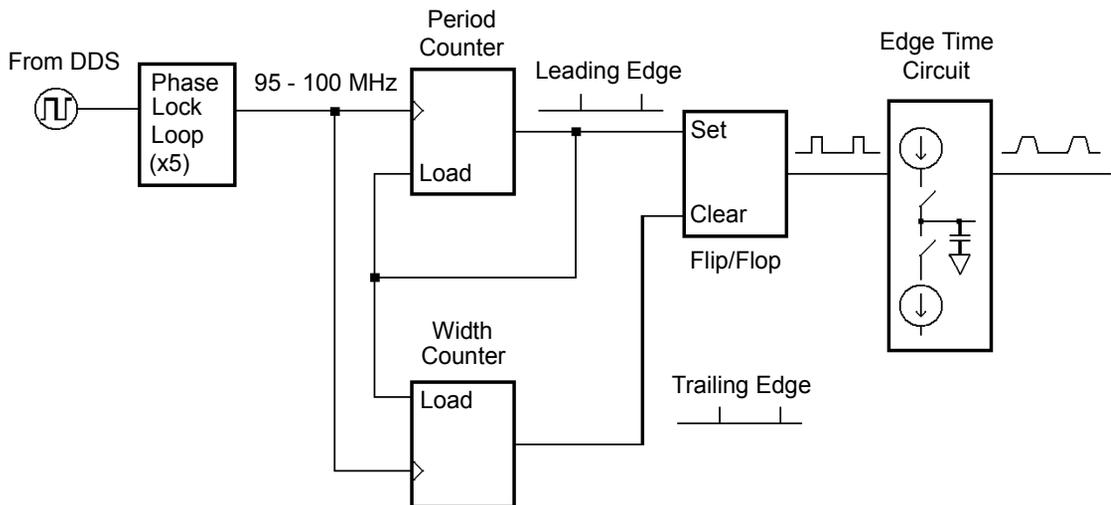
To eliminate distortion due to aliasing at higher frequencies, the 33220A uses a different waveform generation technique to create square waves. Square waveforms are created by routing a DDS-generated sine wave into a comparator. The digital output from the comparator is then used as the basis for the square wave output. The duty cycle of the waveform can be varied by changing the comparator's threshold.



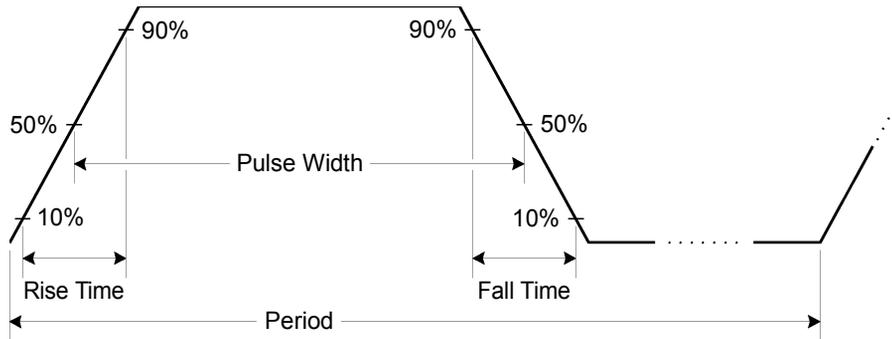
Square Waveform Generation Circuitry

Pulse Waveform Generation

To eliminate distortion due to aliasing at higher frequencies, the Agilent 33220A also uses a different waveform generation technique to create pulse waveforms. For pulse waveform generation, clock cycles are counted to derive both the period and the pulse width. To achieve fine period resolution, the clock frequency is varied from 95 to 100 MHz by a phase lock loop (PLL) circuit (which also multiplies the incoming frequency from the DDS by five). The rising and falling edge times are controlled by a circuit that varies the charging currents in a capacitor. Period, pulse width, and edge time are controlled independently, within certain limits. The pulse waveform generation circuitry is represented in the following block diagram.



Pulse Waveform Generation Circuitry



Pulse Waveform Parameters

Signal Imperfections

For sine waveforms, signal imperfections are easiest to describe and observe in the frequency domain using a spectrum analyzer.

Any component of the output signal which has a different frequency than the fundamental (or “carrier”) is considered to be spurious.

The signal imperfections can be categorized as *harmonic*, *non-harmonic*, or *phase noise* and are specified in “decibels relative to the carrier level” or “dBc”.

Harmonic Imperfections Harmonic components always appear at multiples of the fundamental frequency and are created by non-linearities in the waveform DAC and other elements of the signal path. At low amplitudes, another possible source of harmonic distortion is due to the current flowing through the cable connected to the function generator’s *Sync* output connector. This current can cause a small square-wave voltage drop across the resistance of the cable’s shield and some of this voltage can be imposed on the main signal. If this is a concern for your application, you should remove the cable or disable the *Sync* output connector. If your application requires that you use the *Sync* output connector, you can minimize the effect by terminating the cable in a high impedance load (rather than into a 50Ω load).

Non-Harmonic Imperfections The biggest source of non-harmonic spurious components (called “spurs”) is the waveform DAC. Nonlinearity in the DAC leads to harmonics that are aliased, or “folded back”, into the passband of the function generator. These spurs are most significant when there is a simple fractional relationship between the signal frequency and the function generator’s sampling frequency (50 MHz). For example, at 15 MHz, the DAC produces harmonics at 30 MHz and 45 MHz. These harmonics, which are 20 MHz and 5 MHz from the function generator’s 50 MHz sampling frequency, will appear as spurs at 20 MHz and 5 MHz.

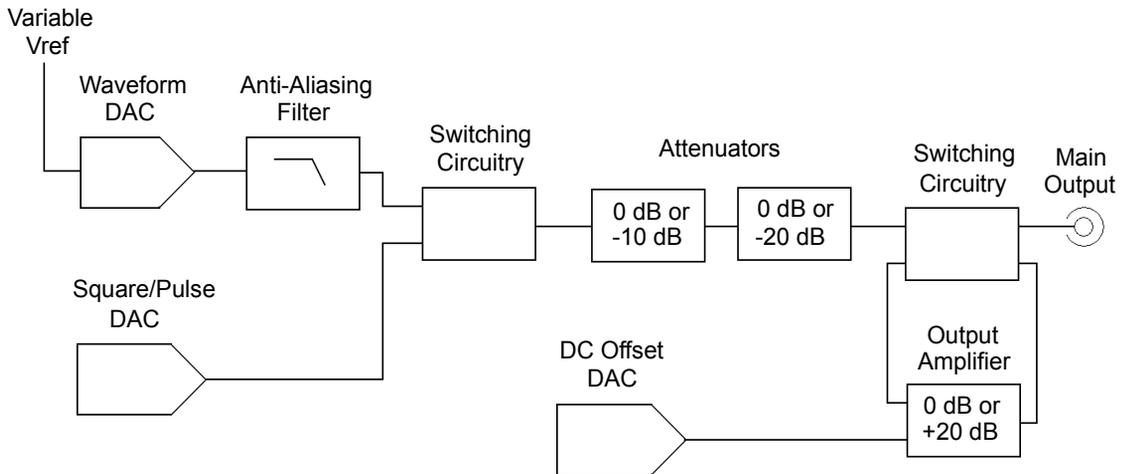
Another source of non-harmonic spurs is the coupling of unrelated signal sources (such as the microprocessor clock) into the output signal. These spurs usually have a constant amplitude (≤ -75 dBm or $112 \mu\text{Vpp}$) regardless of the signal’s amplitude and are most troublesome at signal amplitudes below 100 mVpp. To obtain low amplitudes with minimum spurious content, keep the function generator’s output level relatively high and use an external attenuator if possible.

Phase Noise Phase noise results from small, instantaneous changes in the output frequency (“jitter”). It is seen as an elevation of the apparent noise floor near the fundamental frequency and increases at 6 dBc/octave with the carrier frequency. The 33220A’s phase noise specification represents the amplitude of the noise in a 1 Hz bandwidth, 10 kHz away from a 20-MHz carrier.

Quantization Errors Finite DAC resolution (14 bits) leads to voltage quantization errors. Assuming the errors are uniformly distributed over a range of ± 0.5 least-significant bit (LSB), the equivalent noise level is -86 dBc for a sine wave that uses the full DAC range (16,384 levels). Similarly, finite-length waveform memory leads to phase quantization errors. Treating these errors as low-level phase modulation and assuming a uniform distribution over a range of ± 0.5 LSB, the equivalent noise level is -76 dBc for a sine wave that is 16K samples long. All of the 33220’s standard waveforms use the entire DAC range and are 16K samples in length. Any arbitrary waveforms that use less than the entire DAC range, or that are specified with fewer than 16,384 points, will exhibit proportionally higher relative quantization errors.

Output Amplitude Control

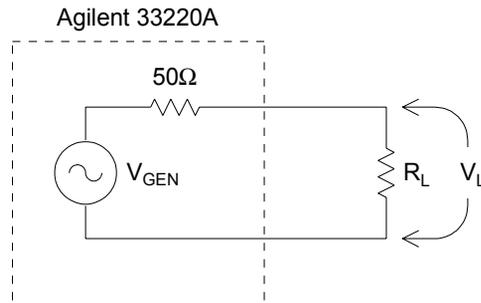
The Agilent 33220A uses a variable reference voltage to control the signal amplitude over a 10 dB range. As shown in the simplified block diagram below, the output of the waveform DAC goes through an anti-aliasing filter. Switching circuitry selects either the waveform output or the output of the separate square/pulse DAC. Two attenuators (-10 dB and -20 dB) are used in various combinations to control the output amplitude in 10-dB steps over a wide range of amplitude values (10 mVpp to 10 Vpp)



Note that the dc offset is summed with the ac signal in the output amplifier. This allows relatively small ac signals to be offset by relatively large dc voltages. For example, you can offset a 100 mVpp signal by almost 5 Vdc (into a 50Ω load).

When changing ranges, the 33220A switches attenuators such that the output voltage never exceeds the current amplitude setting. However, momentary disruptions or "glitches" caused by switching can cause problems in some applications. For this reason, the 33220A incorporates a range hold feature to "freeze" the attenuator and amplifier switches in their current states. However, the amplitude and offset accuracy and resolution (as well as waveform fidelity) may be adversely affected when reducing the amplitude below the expected range change.

As shown below, the 33220A has a fixed series output impedance of 50Ω, forming a voltage divider with the load resistance.

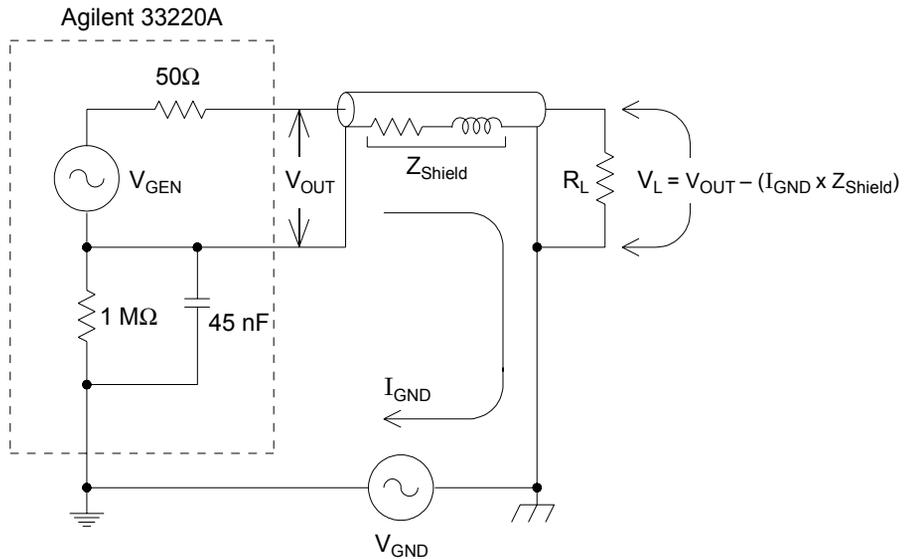


As a convenience, you can specify the load impedance as seen by the function generator and thereby display the correct load voltage. If the actual load impedance is different than the value specified, the displayed amplitude, offset, and high/low levels will be incorrect. Variations in source resistance are measured and taken into account during an instrument calibration. Therefore, the accuracy of the load voltage depends primarily on the accuracy of the load resistance as shown below.

$$\Delta V_L(\%) \cong \frac{50}{R_L + 50} \times \Delta R_L(\%)$$

Ground Loops

The signal-generation portion of the Agilent 33220A is isolated from chassis (earth) ground. This isolation helps to eliminate ground loops in your system and also allows you to reference the output signal to voltages other than ground. The illustration on the following page shows the function generator connected to a load through a coaxial cable. Any difference in ground potentials (V_{GND}) will tend to make current I_{GND} flow in the shield of the cable, thus causing a voltage drop due to the shield's impedance (Z_{Shield}). The resulting voltage drop ($I_{GND} \times Z_{Shield}$) appears as an error in the load voltage. However, since the instrument is isolated, there is a high series impedance (typically 1 MΩ in parallel with 45 nF) in the path to oppose the flow of I_{GND} and thereby minimize this effect.



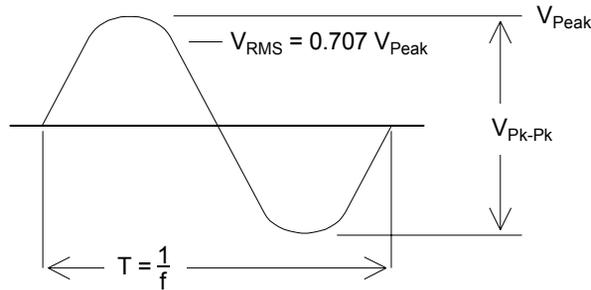
Ground Loop Effects

At frequencies above a few kilohertz, a coaxial cable's shield becomes inductive, rather than resistive, and the cable acts as a transformer. When this happens, it tends to force the shield and center-conductor currents to be equal but opposite. For any voltage drop in the shield due to I_{GND} , there is a similar drop in the center conductor. This is known as the *balun effect* and it reduces ground loops at higher frequencies. Note that lower shield resistance causes the balun effect to become more of a factor at lower frequencies. Therefore, coaxial cables with two or three-braided shields are much better than those with single-braided or foil shields.

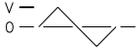
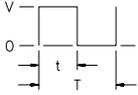
To reduce errors due to ground loops, connect the function generator to the load using a high-quality coaxial cable and ground it at the load through the cable's shield. If possible, make sure the function generator and the load are connected to the same electrical outlet to minimize further differences in ground potential.

Attributes of AC Signals

The most common ac signal is a sine wave. In fact, any periodic signal can be represented as the sum of different sine waves. The magnitude of a sine wave is usually specified by its peak, peak-to-peak, or root-mean-square (RMS) value. All of these measures assume that the waveform has zero offset voltage.



The *peak voltage* of a waveform is the maximum absolute value of all points in the waveform. The *peak-to-peak voltage* is the difference between the maximum and minimum. The *RMS voltage* is found by summing the squares of the voltages of every point in the waveform, dividing the sum by the number of points, and then taking the square root of that quotient. The RMS value of a waveform also represents the one-cycle average power in the signal: $\text{Power} = V_{\text{RMS}}^2 / R_L$. *Crest factor* is the ratio of a signal's peak value to its RMS value and will differ according to waveshape. The table below shows several common waveforms with their respective crest factors and RMS values.

Waveform Shape	Crest Factor (C.F.)	AC RMS	AC+DC RMS
	1.414	$\frac{V}{1.414}$	$\frac{V}{1.414}$
	1.732	$\frac{V}{1.732}$	$\frac{V}{1.732}$
	$\sqrt{\frac{T}{t}}$	$\frac{V}{\text{C.F.}} \times \sqrt{1 - \left(\frac{1}{\text{C.F.}}\right)^2}$	$\frac{V}{\text{C.F.}}$

Chapter 7 Tutorial
Attributes of AC Signals

Note. *If an average-reading voltmeter is used to measure the “DC voltage” of a waveform, the reading may not agree with the DC Offset setting of the function generator. This is because the waveform may have a non-zero average value that would be added to the DC Offset.*

You may occasionally see ac levels specified in “decibels relative to 1 milliwatt” (dBm). Since dBm represents a power level, you will need to know the signal’s RMS voltage and the load resistance in order to make the calculation.

$$\text{dBm} = 10 \times \log_{10}(P/0.001) \quad \text{where } P = V_{\text{RMS}}^2 / R_L$$

For a sine wave into a 50Ω load, the following table relates dBm to voltage.

dBm	RMS Voltage	Peak-to-Peak Voltage
+23.98 dBm	3.54 Vrms	10.00 Vpp
+13.01 dBm	1.00 Vrms	2.828 Vpp
+10.00 dBm	707 mVrms	2.000 Vpp
+6.99 dBm	500 mVrms	1.414 Vpp
0.00 dBm	224 mVrms	632 mVpp
-6.99 dBm	100 mVrms	283 mVpp
-10.00 dBm	70.7 mVrms	200 mVpp
-36.02 dBm	3.54 mVrms	10.0 mVpp

For 75Ω or 600Ω loads, use the following conversions.

$$\begin{aligned} \text{dBm (75}\Omega) &= \text{dBm (50}\Omega) - 1.76 \\ \text{dBm (600}\Omega) &= \text{dBm (50}\Omega) - 10.79 \end{aligned}$$

Modulation

Modulation is the process of modifying a high-frequency signal (called the *carrier signal*) with low-frequency information (called the *modulating signal*). The carrier and modulating signals can have any waveshape, but the carrier is usually a sine waveform.

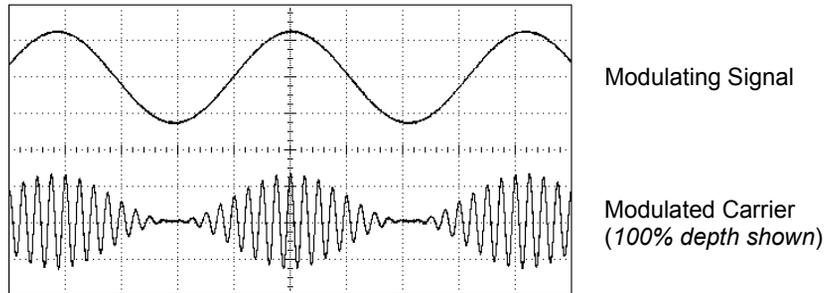
The two most common types of modulation are *amplitude modulation* (AM) and *frequency modulation* (FM). These two forms of modulation modify the carrier's amplitude or frequency, respectively, according to the instantaneous value of the modulating signal. A third type of modulation is *phase modulation* (PM), which is similar to FM except that the phase of the carrier waveform is varied, rather than its frequency. Another type of modulation is *frequency-shift keying* (FSK), where the output frequency "shifts" between two frequencies depending on the state of a digital modulating signal. Finally, pulse width modulation (PWM), is provided for pulse waveforms only. In PWM, the pulse width or duty cycle of the pulse waveform is varied according to the modulating signal.

The function generator will accept an *internal* or *external* modulation source. If you select the *internal* source, the modulated waveform is generated by a secondary DDS synthesizer. If you select the *external* source, the modulated waveform is controlled by the signal level present on the function generator's rear-panel *Modulation In* connector. The external signal is sampled and digitized by an analog-to-digital converter (ADC). With either modulation source, the result is a stream of digital samples representing the modulating waveform.

Note that for FSK, the output frequency is determined by the signal level present on the rear-panel *Trig In* connector.

Modulation

Amplitude Modulation (AM) The function generator implements a form of AM called “double-sideband transmitted carrier.” This is the type of modulation used by most AM radio stations.

*Amplitude Modulation*

The amount of amplitude modulation is called the *modulation depth* which refers to the portion of the amplitude range that will be used by the modulation. For example, a depth setting of 80% causes the amplitude to vary from 10% to 90% of the amplitude setting ($90\% - 10\% = 80\%$) with either an internal or a full-scale ($\pm 5\text{V}$) external modulating signal.

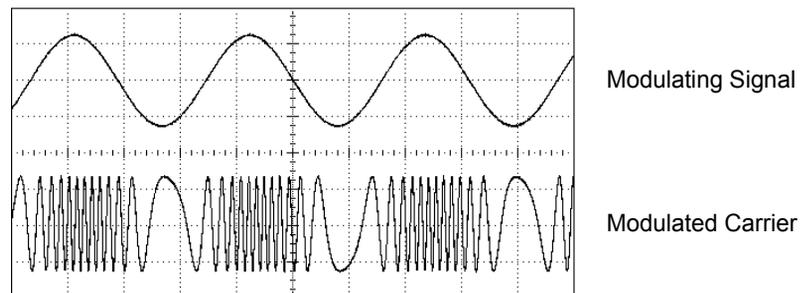
Frequency Modulation (FM) For FM, the function generator uses modulation samples to modify the output frequency of the instrument by changing the content of the PIR (see “*Direct Digital Synthesis*” on page 313). Note that since the rear-panel *Modulation In* connector is decoupled, you can use the 33220A to emulate a voltage-controlled oscillator (VCO).

The variation in frequency of the modulated waveform from the carrier frequency is called the *frequency deviation*. Waveforms with frequency deviations less than 1% of the modulating signal’s bandwidth are referred to as *narrowband FM*. Waveforms with larger deviations are referred to as *wideband FM*. The bandwidth of the modulated signal can be approximated by the following equations.

$$BW \cong 2 \times (\text{Modulating Signal Bandwidth}) \quad \text{For narrowband FM}$$

$$BW \cong 2 \times (\text{Deviation} + \text{Modulating Signal Bandwidth}) \quad \text{For wideband FM}$$

In the United States, commercial FM stations usually have a modulation bandwidth of 15 kHz and deviation of 75 kHz, making them “wideband”. Therefore, the modulated bandwidth is: $2 \times (75 \text{ kHz} + 15 \text{ kHz}) = 180 \text{ kHz}$. Channel spacing is 200 kHz.



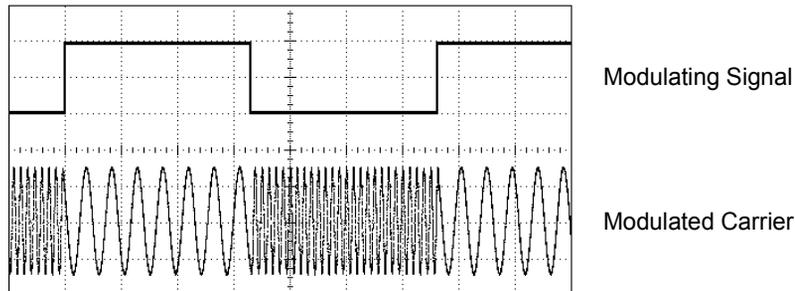
Frequency Modulation

Modulation

Phase Modulation (PM) PM is very similar to FM, but for PM, the *phase* of the carrier waveform is varied, rather than the *frequency*. The variation in phase of the modulated waveform from the carrier waveform is called the *phase deviation*, which can vary from 0 to 360 degrees. Note that since 360 degrees of deviation is equivalent to 0 degrees, the maximum effective deviation is at 180 degrees (the default).

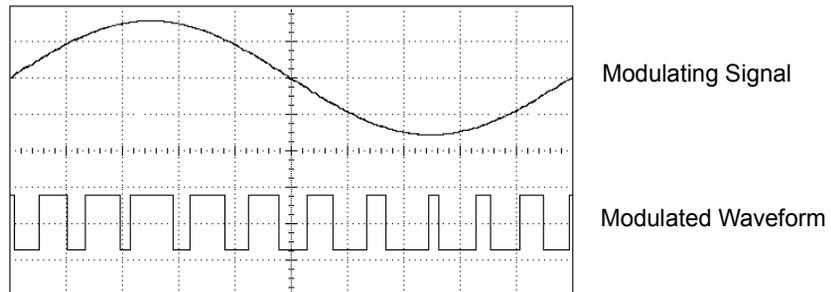
Frequency-Shift Keying (FSK) FSK is similar to FM except the frequency alternates between two preset values. The rate at which the output shifts between the two frequencies (called the “carrier frequency” and the “hop frequency”) is determined by the internal rate generator or the signal level on the rear-panel *Trig In* connector. Frequency changes are instantaneous and phase-continuous.

The internal modulating signal is a square waveform with 50% duty cycle. You can set the internal FSK rate from 2 mHz to 100 kHz.



Frequency-Shift Keying

Pulse Width Modulation (PWM) PWM is used in digital audio applications, motor control circuitry, switching power supplies, and other control applications. The Agilent 33220A provides PWM for pulse waveforms, and PWM is the only type of modulation supported for pulse waveforms. For PWM, the amplitude of the modulating waveform is sampled digitally and used to control the pulse width or duty cycle of the pulse waveform.



Pulse Width Modulation

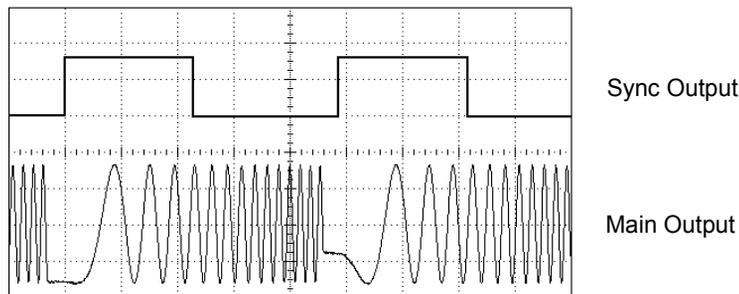
The variation of pulse width in the modulated waveform from the pulse width of the pulse waveform is called the *width deviation*. The deviation can also be expressed in terms of duty cycle (as a percentage referenced to the period of the pulse waveform), which is called the *duty cycle deviation*. In PWM the deviation of width or duty cycle is symmetrical around the pulse width or duty cycle of the original pulse waveform. For example, if you specify a pulse waveform with a 10% duty cycle, and then specify PWM with a duty cycle deviation of 5%, the modulated waveform will have a pulse that varies in duty cycle from 5% to 15% under control of the modulating waveform.

Frequency Sweep

Frequency sweeping is similar to FM but no modulating waveform is used. Instead, the function generator sets the output frequency based on either a linear or logarithmic function. In a *linear* sweep, the output frequency changes in a constant “hertz per second” manner. In a *logarithmic* sweep, the output frequency changes in a constant “octaves per second” or “decades per second” manner. Logarithmic sweeps are useful for covering wide frequency ranges where resolution at low frequencies could be lost with a linear sweep.

You can generate a sweep using an *internal* trigger source or an *external* hardware trigger source. When the *internal* source is selected, the function generator outputs a continuous sweep at a rate determined by the *sweep time* specified. When the *external* source is selected, the function generator will accept a hardware trigger applied to the rear-panel *Trig In* connector. The function generator initiates one sweep each time *Trig In* receives a TTL pulse.

A sweep consists of a finite number of small frequency steps. Since each step takes the same amount of time, longer sweep times result in smaller steps and therefore better resolution. The number of discrete frequency points in the sweep is automatically calculated by the function generator and is based on the *sweep time* you select.



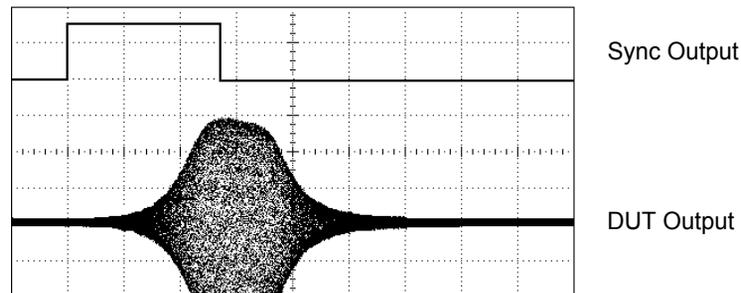
Frequency Sweep

For triggered sweeps, the trigger source can be an external signal, the  key, or a command received from the remote interface. The input for external trigger signals is the rear-panel *Trig In* connector. This connector accepts TTL-compatible levels and is referenced to chassis ground (not floating ground). When not used as an input, the *Trig In*

connector can be configured as an output to enable the 33220A to trigger other instruments at the same time as its internal trigger occurs.

Sync and Marker Signals The output from the front-panel *Sync* connector goes “high” at the beginning of each sweep. If you have *disabled* the Marker function, the Sync signal goes “low” at the midpoint of the sweep. However, if you have *enabled* the Marker function, the Sync signal goes “low” when the output frequency reaches the specified marker frequency. The marker frequency must be between the specified start frequency and stop frequency.

You can use the Marker function to identify a notable frequency in the response of a device under test (DUT) – for example, you may want to identify a resonance. To do this, connect the *Sync* output to one channel of your oscilloscope and connect the DUT output to another channel. Then, trigger the oscilloscope with the rising edge of the Sync signal to position the start frequency on the left side of the screen. Adjust the marker frequency until the falling edge of the Sync signal lines up with the interesting feature in the device’s response. You can then read the frequency from the front-panel display of the 33220A.



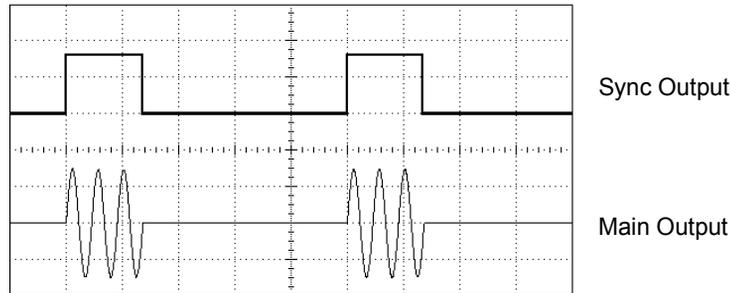
Sweep with Marker at DUT Resonance

Burst

You can configure the function generator to output a waveform with a specified number of cycles, called a *burst*. You can use burst in one of two modes: *N-Cycle Burst* (also called “triggered burst”) or *Gated Burst*.

Burst

N-Cycle Burst An N-Cycle burst consists of a specific number of waveform cycles (1 to 50,000) and is always initiated by a trigger event. You can also set the burst count to “Infinite” which results in a continuous waveform once the function generator is triggered.

***Three-Cycle Burst Waveform***

For bursts, the trigger source can be an external signal, an internal timer, the  key, or a command received from the remote interface. The input for external trigger signals is the rear-panel *Trig In* connector. This connector accepts TTL-compatible levels and is referenced to chassis ground (not floating ground). When not used as an input, the *Trig In* connector can be configured as an output to enable the 33220A to trigger other instruments at the same time as its internal trigger occurs.

An N-Cycle burst always begins and ends at the same point in the waveform, called the *start phase*. A starting phase of 0° corresponds to the beginning of the waveform record and 360° corresponds to the end of the waveform record.

Gated Burst In the Gated burst mode, the output waveform is either “on” or “off” based on the level of the external signal applied to the rear-panel *Trig In* connector. When the gate signal is *true*, the function generator outputs a continuous waveform. When the gate signal goes *false*, the current waveform cycle is completed and then the function generator stops while remaining at the voltage level corresponding to the starting burst phase of the selected waveform. For a noise waveform, the output stops immediately when the gate signal goes false.