

## **NI LabVIEW Modulation Toolkit Help**

#### Version 4.1

August 2008, 370940F-01

The NI LabVIEW Modulation Toolkit contains VIs that can be used with National Instruments hardware or in a simulation environment to generate and analyze analog and digitally modulated signals. The Modulation Toolkit supports ASK, FSK, MSK, PSK, QAM, CPM, PAM, AM, FM, and PM modulation formats, and is capable of IF to I/Q conversion, I/Q visualization, and adding common signal impairments.

For more information about this help file, refer to the following topics:

<u>Conventions</u>

**Related Documentation** 

**Important Information** 

Technical Support and Professional Services

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## **Related Documentation**

The following documents contain information that you might find helpful as you use this help file:

#### **NI Documents**

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**Note** For instructions on printing this help file, click the **Search** tab and type print.

- NI LabVIEW Modulation Toolkit Readme, installed at Start»All Programs»National Instruments»Modulation»LabVIEW Support»Modulation Readme. This file contains important information about modulation VIs.
- Modulation Toolkit example VIs, installed at Start»All Programs»National Instruments»Modulation»LabVIEW Support»Modulation Examples Folder. This folder contains LabVIEW examples which employ Modulation Toolkit VIs. You can also browse all installed examples and their descriptions by opening LabVIEW, pulling down the Help menu, and navigating to Toolkits and Modules»Modulation.
- *NI Spectral Measurements Toolkit Readme*, installed at **Start»All Programs»National Instruments»Spectral Measurements»Spe** this help file contains important information about modulation and Spectral Measurements Toolkit VIs.
- Spectral Measurements Toolkit User Guide, installed in PDF format at Start»All
- Programs»National Instruments»Spectral Measurements.
  NI LabVIEW Spectral Measurements Toolkit Help, installed at
  - Start»All Programs»National Instruments»Spectral Measurements»Lak
- NI-SCOPE VI Reference Help, contained in the NI High-Speed Digitizers Help, installed at Start»All Programs»National Instruments»NI-SCOPE»Documentation.
- *NI RF Vector Signal Analyzers Help*, available at **Start»All Programs»National Instruments»NI-RFSA»Documentation**.
- *NI RF Vector Signal Analyzers Getting Started Guide*, printed and installed in PDF format at **Start»All Programs»National Instruments»NI-RFSA»Documentation**.
- *MAX Remote Systems Help*, available in Measurement & Automation Explorer (MAX) by selecting **Help\*HelpTopics\*Remote Systems**.
- LabVIEW Real-Time Module User Manual, available at

#### ni.com/manuals.

You can download PDF versions of the *Spectral Measurements Toolkit User Guide* and the *NI RF Vector Signal Analyzers Getting Started Guide* at <u>ni.com/manuals</u> (link opens in a new window).

The following resources contain information about concepts related to the Modulation Toolkit.

- Note The following resources offer useful background information about the general concepts discussed in this documentation. These resources are provided for general informational purposes only and are not affiliated, sponsored, or endorsed by National Instruments. The content of these resources is not a representation of, may not correspond to, and does not imply current or future functionality in any other National Instruments product.
  - Leiner, Bernhard M.J. *LDPC Codes—a brief Tutorial* http://users.tkk.fi/pat/coding/essays/ldpc.pdf. 2005.
  - Lin, S., and DJ Costello, Jr. *Error Control Coding: Fundamentals and Applications.* Englewood Cliffs: Prentice-Hall, 1983.
  - McEliece, Robert J. *Finite Fields for Computer Scientists and Engineers (The Kluwer International Series in Engineering and Computer Science)*. New York: Springer Publishers, 1986.
  - Oerder, Martin, and Heinrich Mayer. "Digital Filter and Square Timing Recovery." *IEEE Transactions on Communications* 36 (5): 1988.
  - Premji, Al-Nasir, and Desmond P Taylor. "Receiver Structures for Multi-h Signaling Formats." *IEEE Transactions on Communications* 35(4): 1987.
  - Richardson, Thomas J., and Rüdiger L. Urbanke. "Efficient Encoding of Low-Density Parity-Check Codes." *IEEE Transactions on Information Theory* 47(2): 2001.
  - Shokrollahi, Amin. *LDPC Codes: An Introduction* http://www.ipm.ac.ir/IPM/homepage/Amin2.pdf. 2003.
  - Smith, Steven W. *The Scientists and Engineers Guide to Digital Signal Processing*. California Technical Publishing, 1997.
  - Press, William H., ed., and Teukolsky, Saul A., ed. *Numerical Recipes in C: The Art of Scientific Computing.* 2nd ed.

Cambridge: Cambridge University Press, 1992.

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- Sharon, E., S. Litsyn, and J. Goldberger. "An Efficient Message Passing Schedule for LDPC Decoding." *Proceedings of the 23rd IEEE Convention of Electrical and Electronics Engineers in Israel* 2004.
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- Sun, Jian. *An Introduction to Low Density Parity Check (LDPC) Codes.* Wireless Communication Research Laboratory, Lane Dept. of Comp. Sci. and Elec. Engr., West Virginia University.
- Wicker, Stephen B. *Error Control Systems for Digital Communications and Storage*. Englewood Cliffs: Prentice Hall, 1995.

## **Using Help**

<u>Conventions</u> <u>Navigating Help</u> <u>Searching Help</u> <u>Printing Help File Topics</u>

## Conventions

P

This help file uses the following conventions:

- < > Angle brackets that contain numbers separated by an ellipsis represent a range of values associated with a bit or signal name—for example, DBIO<3..0>.
- [] Square brackets enclose optional items—for example, [response].
- The » symbol leads you through nested menu items and dialog box options to a final action. The sequence File»Page Setup»Options directs you to pull down the File menu, select the Page Setup item, and select Options from the last dialog box.
  - This icon denotes a tip, which alerts you to advisory information.
- This icon denotes a note, which alerts you to important information.
- This icon denotes a caution, which advises you of precautions to take to avoid injury, data loss, or a system crash.
- **bold** Bold text denotes items that you must select or click on in the software, such as menu items and dialog box options. Bold text also denotes parameter and cluster names, emphasis, or an introduction to a key concept.
- green Underlined text in this color denotes a link to a help topic, help file, or Web address.
- *italic* Italic text denotes variables or cross references. This font also denotes text that is a placeholder for a word or value that you must supply.
- monospace Text in this font denotes text or characters that you should enter from the keyboard, sections of code, programming examples, and syntax examples. This font is also used for the proper names of disk drives, paths, directories, programs, subprograms, subroutines, device names, functions, operations, variables, filenames and extensions,

and code excerpts.

# **Navigating Help (Windows Only)**

To navigate this help file, use the **Contents**, **Index**, and **Search** tabs to the left of this window or use the following toolbar buttons located above the tabs:

- **Hide**—Hides the navigation pane from view.
- Locate—Locates the currently displayed topic in the Contents tab, allowing you to view related topics.
- **Back**—Displays the previously viewed topic.
- Forward—Displays the topic you viewed before clicking the **Back** button.
- **Options**—Displays a list of commands and viewing options for the help file.

# **Printing Help File Topics (Windows Only)**

Complete the following steps to print an entire book from the **Contents** tab:

- 1. Right-click the book.
- 2. Select **Print** from the shortcut menu to display the **Print Topics** dialog box.
- 3. Select the **Print the selected heading and all subtopics** option.
  - Note Select Print the selected topic if you want to print the single topic you have selected in the **Contents** tab.
- 4. Click the **OK** button.

### **Printing PDF Documents**

This help file may contain links to PDF documents. To print PDF documents, click the print button located on the Adobe Acrobat Viewer toolbar.

## **Searching Help (Windows Only)**

Use the **Search** tab to the left of this window to locate content in this help file. If you want to search for words in a certain order, such as "related documentation," add quotation marks around the search words as shown in the example. Searching for terms on the **Search** tab allows you to quickly locate specific information and information in topics that are not included on the **Contents** tab.

### Wildcards

You also can search using asterisk (\*) or question mark (?) wildcards. Use the asterisk wildcard to return topics that contain a certain string. For example, a search for "prog\*" lists topics that contain the words "program," "programmatically," "progress," and so on.

Use the question mark wildcard as a substitute for a single character in a search term. For example, "?ext" lists topics that contain the words "next," "text," and so on.



**Note** Wildcard searching will not work on Simplified Chinese, Traditional Chinese, Japanese, and Korean systems.

### **Nested Expressions**

Use nested expressions to combine searches to further refine a search. You can use Boolean expressions and wildcards in a nested expression. For example, "example AND (program OR VI)" lists topics that contain "example program" or "example VI." You cannot nest expressions more than five levels.

### **Boolean Expressions**

Click the **•** button to add Boolean expressions to a search. The following Boolean operators are available:

- **AND** (default)—Returns topics that contain both search terms. You do not need to specify this operator unless you are using nested expressions.
- **OR**—Returns topics that contain either the first or second term.
- **NOT**—Returns topics that contain the first term without the second term.
- **NEAR**—Returns topics that contain both terms within eight words of each other.

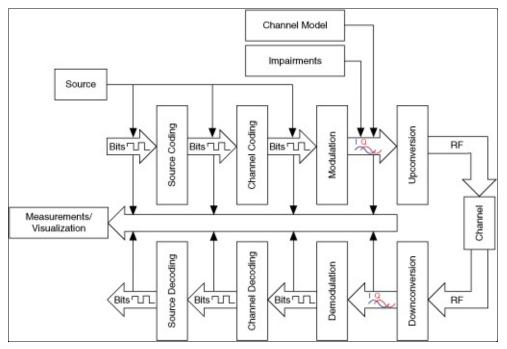
### **Search Options**

Use the following checkboxes on the **Search** tab to customize a search:

- **Search previous results**—Narrows the results from a search that returned too many topics. You must remove the checkmark from this checkbox to search all topics.
- Match similar words—Broadens a search to return topics that contain words similar to the search terms. For example, a search for "program" lists topics that include the words "programs," "programming," and so on.
- Search titles only—Searches only in the titles of topics.

## **Digital Communication System Fundamentals**

A typical digital communication system includes a transmitter, a receiver, and a communication channel. The following figure illustrates the general components of a digital communication system. The transmitter contains blocks for source and channel coding, modulation, simulating real-world signal impairments, and upconversion. The receiver includes blocks for downconversion, matched filtering, equalization, demodulation, and channel and source decoding. You can quantify the results using tools for measurement and visualization.



# Decoding

Data decoding involves removing redundant bits from the sequence and correcting for any errors that might have happened during transmission. The signal decoding process is usually more complicated than the encoding process and can be very computationally intensive. However efficient decoding schemes have been developed over the years —one example is the Viterbi decoding algorithm, which is used to decode convolutionally encoded data.

The following VIs are used in channel decoding:

- MT Hamming Decoder
- MT Golay Decoder
- <u>MT Reed-Solomon Decoder</u>
- MT BCH Decoder
- MT Convolutional Decoder
- <u>MT Despread Symbols</u>
- <u>MT LDPC Decoder</u>

## Demodulation

The downconverted signal undergoes a <u>demodulation</u> process. This step is the opposite of <u>modulation</u> and refers to the process required to extract the original <u>information signal</u> from the <u>modulated signal</u>.

The process of digital demodulation involves <u>matched filtering</u>, <u>symbol</u> <u>timing extraction</u>, and symbol synchronization followed by frequency offset correction. The frequency-offset-corrected signal is decimated down to symbol-spaced data, which is then mapped back to a recovered bit stream.

The following VIs are used in analog demodulation:

- MT Demodulate AM
- <u>MT Demodulate FM</u>
- <u>MT Demodulate PM</u>

Digital demodulation returns the time-aligned demodulated waveform, the demodulated information bit stream, and measurement results obtained during demodulation. The following VIs are used in digital demodulation:

- MT Demodulate ASK
- MT Demodulate FSK
- <u>MT Demodulate MSK</u>
- MT Demodulate PAM
- <u>MT Demodulate PSK</u>
- <u>MT Demodulate QAM</u>
- MT Demodulate CPM

Note Use the following VIs if your application requires only the demodulated bit stream.

- <u>MT Detect ASK</u>
- MT Detect FSK
- <u>MT Detect MSK</u>
- <u>MT Detect PAM</u>
- MT Detect PSK
- <u>MT Detect QAM</u>
- <u>MT Detect CPM</u>

### Downconversion

The first step in the demodulation process is downconversion from a real passband waveform to a complex I/Q baseband waveform. This process involves mixing the real-valued passband waveform with a locally generated carrier tone, followed by lowpass filtering to generate the I/Q baseband waveform.

Use the <u>MT Downconvert Passband</u> VI to downconvert waveforms.

# Encoding

A data source generates the <u>information signal</u> sent to a particular receiver. This signal may be either an analog signal, such as speech, or a digital signal, such as a binary data sequence. The information signal is typically a <u>baseband</u> signal represented by a voltage level.

The Source Coding block typically involves data compression. For example, the ATSC standard for digital video broadcast (DVB) specifies MPEGII encoding for the transmitted image. A-law, Mu-law, JPEG, A-87.6 are examples of other compression algorithms commonly used in source coding.

The Channel Coding block typically involves adding redundant bits to the data stream to increase the receiver's immunity to noise and interference in the channel. The output of the Channel Coding block is a series of 0s and 1s. Among the most popular error-correcting schemes are block and convolutional coding.

Note The Modulation Toolkit does not currently support source coding. You can use LabVIEW VIs and primitives to apply source coding in your application.

The Modulation Toolkit provides the following VIs for channel coding:

- MT Hamming Encoder
- MT Golay Encoder
- <u>MT Reed Solomon Encoder</u>
- <u>MT BCH Encoder</u>
- MT Convolutional Encoder
- <u>MT Spread Symbols</u>
- <u>MT LDPC Encoder</u>

## **Direct Sequence Spread Spectrum (DSSS)**

*Direct sequence spread spectrum* (DSSS) is a process by which data is transmitted using a higher bandwidth signal that is demanded by the data rate. Using DSSS allows multiple channels to occupy the same bandwidth, thus mitigating interference from other users at the expense of bandwidth expansion.

DSSS spreads each bit of signal data at the transmitter into *L* chips using a pseudorandom *L*-chip spreading code called a *code word*. The length *L* of the pseudorandom spreading code is also known as the *bandwidth expansion factor* because the chips are transmitted at a rate equal to  $L \times bit$  rate of the data. The spreading code appears random to all receivers except the intended one, which uses the knowledge of the spreading code to demodulate and recover the transmitted information. Thus multiple channels can occupy the same portion of the frequency spectrum by using code words that have little or no correlation with one another, and little or no autocorrelation for any shift other than zero.

Mathematically, a DSSS signal is described by

$$y(t) = \sum_{n=-\infty}^{\infty} \sum_{m=0}^{L-1} a_n c_m g \left( \tau - n\tau - m\tau_c \right)$$

where

y(T) is the transmitted DSSS signal

g(T) is the pulse-shaping signal of duration  $T_c$ 

 $\{a_n\}$  is the n<sup>th</sup> information bearing symbol

 $\{c_m\}$  is the m<sup>th</sup> element of the *L*-long pseudorandom spreading code

(also known as the chip sequence)

 $T_c$  is the chip period

 $T=L\times T_c$  is the symbol period

# Low-Density Parity Check (LDPC) Encoding

Low-density parity check (LDPC) is a linear error-correcting code that uses a parity check matrix that provides only a few 1s with respect to a much larger number of 0s. The main advantage of the parity check matrix is that it provides a performance that is close to the capacity of many different channels and linear time complex algorithms for decoding. Furthermore, parity check matrices are suited for implementations that make heavy use of parallelism.

An LDPC code is a block code that has a parity check matrix *H*, every row and column of which is sparse. A Regular Gallager Code is a LDPC code in which every column of H has some weight, *j*, and every row has some weight, *k*. Regular Gallager codes are constructed at random subject to these constraints.

For example, if

- the number of ones in each column (j) = 3
- the number of ones in each row (k) = 6
- the number of columns (*n*) = 12
- the number of rows (m) = 6 (because  $m = n^* j/k$ )
- the rate of (n, j, k) LDPC Code is  $R \ge 1$  (j/k)

Then

If the number of 1s per column or row is not constant, the code is an irregular LDPC code. Usually, irregular LDPC codes outperform regular LDPC codes.

Refer to the following resources for more information about the algorithms and methods used in LDPC coding:

- Bernhard M. J. Leiner, "LDPC codes a Brief Tutorial," April 2005. http://users.tkk.fi/pat/coding/essays/ldpc.pdf
- Shokrollahi, Amin. LDPC Codes: An Introduction. Digital Fountain, Inc. Fremont: 2004.

http://www.ipm.ac.ir/IPM/homepage/Amin2.pdf

- Richardson, Thomas J. and Rüdiger L. Urbanke. "Efficient Encoding of Low-Density Parity-Check Codes." *IEEE Transactions on Information Theory* 47(2): 2001.
- The flooding decoding algorithm is according to following reference : Sun, Jian. "An Introduction to Low Density Parity Check (LDPC) Codes." *WCRL Seminar Series* Wireless Communication Research Laboratory,West Virginia University: 2003.
- The serial decoding algorithm is according to following reference : Sharon, E. et al. "An Efficient Message Passing Schedule for LDPC Decoding." 2004 23rd IEEE Convention of Electrical and Electronics Engineers in Israel 23: 2004.

# Equalization

The Modulation Toolkit employs an adaptive feed-forward equalizer, which implies that the equalizer taps continuously adapts its coefficients to compensate for the action of the channel filter. The adaptive feedforward equalizer uses a feed-forward adaptive least-mean-squared (LMS) algorithm to adjust the equalizer taps. At the start of the equalization process, you must supply training bits to train the equalizer. After training, the equalizer switches to decision-directed feedback mode, where the equalizer trains itself based on its own decisions.

The following are types of digital feedforward equalization VIs:

- <u>ASK</u>
- <u>PAM</u>
- <u>PSK</u>
- <u>QAM</u>

## Filtering

In a digital communication system, digital information can be sent on a carrier through changes in its fundamental characteristics such as phase, frequency, and amplitude. In a physical channel, these transitions can be smoothed, depending on the filters implemented during transmission. In fact, filters play an important part in a communications channel because they can eliminate spectral leakage, reduce channel width, and eliminate adjacent symbol interference known as inter-symbol interference (ISI).

The matched filter is as important as the pulse-shaping filter. Though the pulse-shaping filter generates signals such that each symbol period does not overlap, the matched filter is important because it filters out the signal reflections that occur in the transmission process. Because a direct-path signal arrives at the receiver before a reflected signal does, it is possible for the reflected signal to overlap with a subsequent symbol period. The matched filter reduces this affect by attenuating the beginning and ending of each symbol period. Thus, it can reduce ISI.

The Modulation Toolkit provides the following types of filters:

- Raised cosine
- Root-raised cosine
- Gaussian pulse-shaping and matched filters

## **Filter Delay**

Modulation Toolkit uses finite impulse response (FIR) filters for different operations like pulse-shaping, matched, and downconversion filtering. For such filters, the output signal is related to the input signal as shown by the following equation:

 $y[n] = b_0 x[n] + b_1 x[n-1] + \dots + b_P x[n-P]$ 

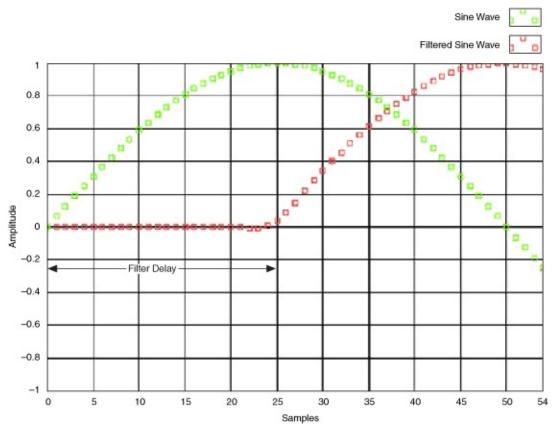
where P is the filter order

*x*[*n*] is the input signal

y[n] is the output signal

 $b_i$  are the filter coefficients

The initial state for all samples in an FIR filter is 0. The filter output until the first input sample reaches the middle tap (the first causal sample) is called the *transient response*, or *filter delay*. Given an FIR filter which has N taps, the delay is (N - 1)/2 samples. This relationship is illustrated in the following figure, where a sine wave is filtered by an FIR filter with 50 taps.



### Using the Reset? Parameter in Iterative Modulator/Demodulator Operation

In the Modulation Toolkit, digital modulation VIs initially perform mapping of the input bits onto symbols and subsequently apply a pulse-shaping filter to these symbols. The digital demodulation VIs perform matched filtering and buffering on the recovered complex waveform. Some useful samples are held in these filters.

In an iterative operation where the modulator/demodulator is called repeatedly inside a loop to operate on blocks of bits/symbols (all of which are part of the same message), two options are possible:

- reset? is set to TRUE on every iteration—The filters in the modulator/demodulator are initialized with zeros on every iteration of the loop, and the returned output data is shortened by (*N* - 1)/2 symbols on all iterations.
- **reset?** is set to TRUE on the first call and to FALSE on subsequent iterations—The filters in the modulator/demodulator are initialized with zeros on the first iteration of the loop. On subsequent iterations of the loop, the VI uses state information from previous iterations to initialize the filter.

#### **Recovering Samples in Single Shot Operations**

In single-shot operations for modulators and demodulators, the filter delay is truncated before the signal is generated because these samples are not valid. Some samples at the end of the block do not appear at the modulator/demodulator output, and hence appear to have been lost.

You can recover these samples by sending extra samples to the modulator/demodulator. To determine how many extra samples you must added, use the following guidelines:

• **For Modulation VIs**—Let *L* = pulse-shaping filter length, *m* = number of samples per symbol, and *M* = modulation order. The number of bits to be added to the input bit stream is given by the following formula:

$$N = (L - 1) \quad \frac{\log_2 M}{m}$$

• For Demodulation VIs—The demodulation VIs use filters during

matched filtering. Let L be the length of the matched filter. The number of samples to be added to the input signal prior to filtering is given by the following formula:

$$N = \frac{L-1}{2}$$

The *N* extra samples are obtained by repeating the last sample value of the input signal *N* times to ensure signal continuity.

#### Working with Filter Delay in Modulation Toolkit 4.1

#### Modulation

The modulation VIs in Modulation Toolkit 4.1 contain a new parameter, **flush buffers?**, that allows you to flush out samples that have been delayed in the pulse-shaping filter. When this parameter is set to TRUE, the VI internally pads the input data to recover any lost data. Set this parameter to TRUE for single-shot applications and during the last iteration of a continuous operation application.

### Demodulation

Modulation Toolkit 4.1 provides two methods for demodulation: demodulation VIs and detector VIs. The demodulation VIs contain the same behavior as in Modulation Toolkit 4.0, that is, they allow you to recover the time-aligned demodulated waveform, the demodulated information bit stream, and measurement results obtained during demodulation. The detector VIs return only the demodulated bit stream. These new VIs contain the **flush buffers?** parameter that pads the incoming samples and forces out the samples delayed because of the FIR filters used in the demodulation algorithms.

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**Note** Setting **flush buffers?** to TRUE destroys the internal states of the algorithms such that you will be unable to perform continuous processing on the signal during subsequent iterations. If **flush buffers?** is set to TRUE, you must set **reset?** to TRUE on the subsequent iteration.

## Impairments

All transmission media (including wireless, fiber optic, and copper) introduce some form of distortion to the original signal. Different types of channel models have been developed to mathematically represent such real-world distortions. The Modulation Toolkit can generate a modulated message signal and optionally add noise, impairments, and channel models.

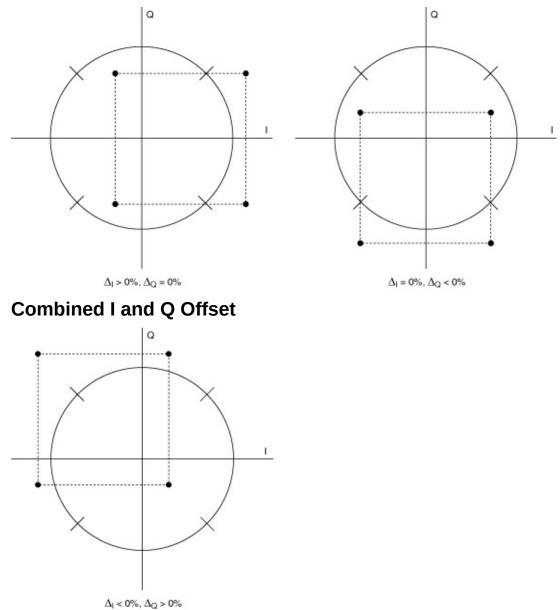
The Modulation Toolkit can add the following types of impairments:

- <u>AWGN</u>
- <u>MultiTone</u>
- <u>I/Q Impairments</u>
- Phase Noise
- Generate Fading Profile
- Fading Profile

## **DC Offset**

*DC offset* is a complex signal impairment that shifts the locus of ideal symbol coordinates off-center in the I/Q plane. A DC offset can be added to the baseband I component, the Q component, or both. The DC offset can be either positive or negative, with the sign indicating the direction of the shift. DC offset is expressed as a percentage of full scale, where "full scale" (fs) is the amplitude of the baseband <u>quadrature modulation</u> (QM) waveform.

#### Individual I and Q Offsets



### **Frequency Selective Fading**

You can use frequency-selective fading to model discrete multipath wireless channels that are dynamic in both space and time. The inputoutput relationship for a discrete multipath model is given by the following equation:

$$y(t) = \sum_{k=1}^{N(t)} a_k(t) \times (t - \tau_k(t))$$

where y(t) is the received signal

 $a_k(t)$  is the complex path attenuation, which is modeled as a random process with a probability distribution that is Rayleigh or Rician for k = 1, N(t)

N(t) is the number of paths in the multipath channel

 $\tau$ (*t*) is their corresponding delays

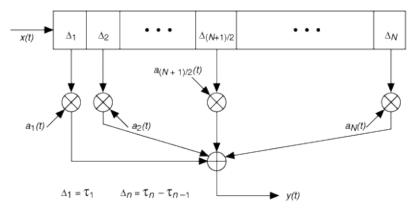
t is time

The Rayleigh fading profile can be implemented using Jakes or Gans models. If the channel model has a line-of-sight path, the envelope of the dominant path (that is, the first path) is Rician distributed. Rician profiles also can be implemented using Jakes or Gans models.

If we assume that the number of multipath components and the delay structure vary slowly compared to the variations in  $a_k$  (*t*), the previous relationship can be rewritten as the following equation:

$$y(t) = \sum_{k=1}^{N} a_k(t) \times (t - \tau_k)$$

This type of system is a linear time variant (LTV) system and can be implemented using a tapped delay line structure as illustrated in the following figure:



The selective fading profile instances of the <u>MT Generate Fading Profile</u> VI generate Rayleigh or Rician distributed fading profiles for the specified **number of paths** based on the Jakes or Gans model. The generated **fading profile** is a two-dimensional array in which the number of rows is equal to the **number of paths**, and the number of columns is equal to the **profile length**. The generated profile is passed to the <u>MT Apply Selective</u> <u>Fading Profile</u> VI.

## **Phase Noise**

*Phase noise* refers to noise in a <u>carrier</u> signal due to phase and frequency <u>modulation</u> in the signal. Phase noise is normally very close to the carrier and is measured in decibels relative to the carrier frequency (dBc).

Phase noise is expressed as a function of power spectral density and frequency. In a 1 Hz bandwidth, phase noise is given by

 $\mathfrak{L}^{(f)} = 10\log[0.5(S \Phi(f))]\Phi$ 

where  $S \Phi(f)$  is the spectral density of phase fluctuations.

### Measurement

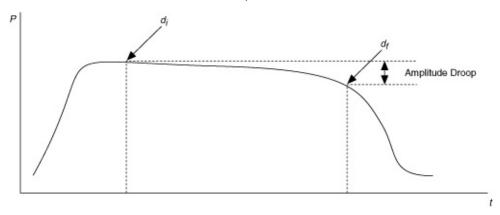
Communications engineers use different tools to evaluate how well the transmitted data was received. <u>Bit error rate (BER)</u> and <u>modulation error</u> <u>ratio (MER)</u> are common metrics for this job.

The following list illustrates the types of measurements that can be performed with Modulation Toolkit VIs.

- Calculate Bit Error Rate (BER) After Trigger:
  - BER
  - Accumulated BER
  - Trigger Found Index
- Digital Demodulation:
  - <u>QAM</u>, <u>ASK</u>, <u>PAM</u>, <u>PSK</u>: Frequency Offset, Frequency Drift, Phase Offset
  - <u>FSK</u>: Frequency Offset, Frequency Drift, Deviation Error, FSK Error
  - MSK: Frequency Offset, Frequency Drift
- <u>Measure Quadrature Impairments:</u>
  - <u>I/Q Gain Imbalance</u>, <u>Quadrature Skew</u>, Magnitude Error, <u>EVM</u>, <u>Phase Error</u>, <u>MER</u>
  - DC Offsets: I, Q, Origin
- <u>Measure Rho</u> (ρ)
- Measure Burst Timing
  - Amplitude Droop, Crest Factor

# **Amplitude Droop**

Amplitude droop, measured in dB, is a measure of the amount that the signal power falls from the start of a specified measurement window  $(d_i)$  to the end of that window  $(d_f)$ .



# **Bit Error Rate (BER)**

*Bit error rate (BER)* is the ratio of erroneous bits to total bits transmitted, received, or processed over some stipulated period. Transmission BER expresses the number of erroneous bits received divided by the total number of bits transmitted. Information BER expresses the number of erroneous decoded (corrected) bits divided by the total number of decoded (corrected) bits.

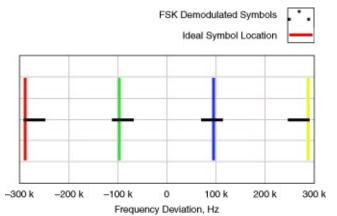
# **Deviation Error**

For an M-FSK system, the average *deviation error* is defined as the average spread magnitude of the FSK demodulated symbol spaced waveform around the ideal symbol (<u>frequency</u>) locations.

Mathematically, the deviation error is defined as

$$f_{dev} = \frac{1}{M} \left| \sum_{i=0}^{M-1} \left( f_{ideal,i} - \left\langle f_{actual,i} \right\rangle \right) \right|$$

where *M* is the FSK modulation format,  $f_{ideal,i}$  is the ideal symbol location at the FSK frequency corresponding to location *i*, and  $< f_{actual,i} >$  is the mean value of the demodulated symbols at location *i*.

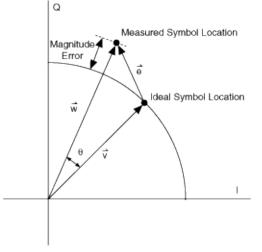


In the preceding figure, the horizontal lines represent the actual FSK demodulated symbol locations, and the vertical lines represent the ideal symbol locations for a 4-FSK format.

# Error Vector Magnitude (EVM)

*Error vector magnitude* (EVM) is a measurement of demodulator performance in the presence of impairments. The soft symbol decisions obtained after decimating the recovered waveform at the demodulator output are compared against the ideal symbol locations. The root mean square (RMS) EVM and phase error are then used in determining the EVM measurement over a window of *N* demodulated symbols.

As shown in the following figure, the symbol decision generated by the demodulator is given by  $\vec{*}$ . However, the ideal symbol location (using the symbol map) is given by  $\vec{*}$ . Therefore, the resulting *error vector* is the difference between the actual measured and ideal symbol vectors given by  $\vec{*} = \vec{*} - \vec{*}$ . The error vector  $\vec{*}$  for a received symbol is graphically represented by the following figure:



#### where

is the ideal symbol vector
 is the measured symbol vector
 is the magnitude error
 θ is the phase error

- $\vec{a} = \vec{w} \vec{v}$  is the error vector
- ਡ/ず is the EVM

EVM quantifies, but does not necessarily reveal the nature of the impairment. To remove the dependence on system gain distribution, EVM is normalized by  $|\underline{v}|$ , which is expressed as a percentage. Analytically, RMS EVM over a measurement window of *N* symbols is defined as

EVM = 
$$\frac{\sqrt{\frac{1}{N}\sum_{j=1}^{N} [(I_j - \tilde{I}_j)^2 + (Q_j - \tilde{Q}_j)^2]}}{|\underline{v}_{max}|}$$

where

 $I_j$  is the I component of the *j*-th symbol received

 $\dot{Q}_j$  is the Q component of the *j*-th symbol received

 $\tilde{I}_{j}$  is the ideal I component of the *j*-th symbol received

 $\tilde{Q}_j$  is the ideal Q component of the *j*-th symbol received

EVM is related to the modulation error ratio (MER) and  $\rho$ . EVM and MER have a one-to-one relationship. EVM measures the vector difference between the measured and ideal signals, while  $\rho$  measures the correlation between the two signals.

# **Frequency Deviation**

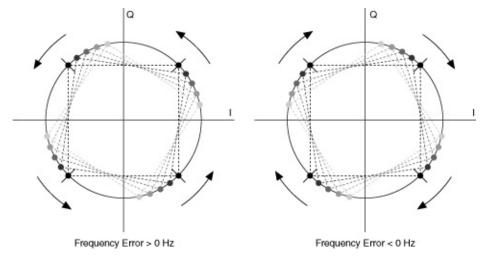
*Frequency deviation* quantifies the amount that a <u>frequency</u> differs from its specified value, as when measuring how much an oscillator frequency deviates from its nominal frequency.

In <u>frequency modulation</u>, frequency deviation refers to the maximum absolute difference, during a specified period, between the instantaneous frequency of the <u>modulated wave</u> and the <u>carrier</u> frequency.

# **Frequency Error, Quadrature Modulation (QM)**

In <u>quadrature-modulated</u> (QM) systems, *frequency error* refers to the difference between the specified carrier frequency and the actual measured carrier frequency.

At <u>baseband</u> frequencies, frequency error in QM systems is a complex signal impairment that manifests itself as a rotation of the locus of symbol coordinates about the I/Q plane. A fixed frequency offset appears to be a rotation at a constant angular velocity. Frequency error can be either positive or negative in sign, indicating the direction of the rotation.



# **Frequency Shift-Keying (FSK) Error**

For an *M*-ary <u>FSK</u> system, the average FSK error is defined as the average root mean squared (RMS) spread of the FSK demodulated symbol spaced waveform around the ideal symbol (frequency) locations.

Mathematically, the FSK error is defined as

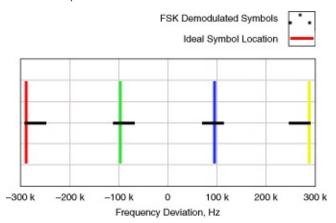
$$f_{dev} = \frac{1}{M} \left[ \sum_{i=0}^{M-1} \sqrt{\left\langle \left(f_{ideal,i} - f_{actual,i}\right)^2 \right\rangle} \right]$$

where

M is the FSK modulation format

 $f_{ideal,i}$  is the ideal symbol location at the FSK frequency corresponding to location *i* 

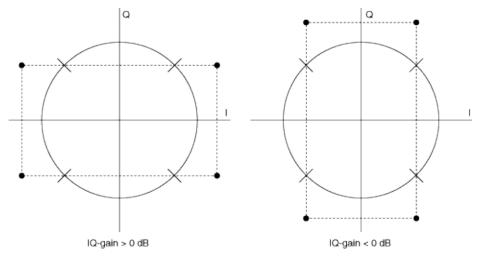
 $f_{actual,i}$  represents the value of the demodulated symbols at location *i* 



In the preceding figure, the points represent the actual FSK demodulated symbol locations, and the vertical lines represent the ideal symbol locations for a 4-FSK format.

### I/Q Gain Imbalance

*I/Q gain imbalance* refers to a difference in scaling between the I and Q components of I/Q data. When expressed in dB, I/Q gain imbalance can be either positive or negative, with the sign indicating which component has been impaired.



# Magnitude Error, Frequency Shift-Keying (FSK)

In FSK-modulated systems, *magnitude error* refers to the difference between the ideal magnitude of the I/Q signal and the actual measured magnitude on a symbol-by-symbol basis. For 2-FSK systems, for example, the magnitude error  $\varepsilon_m$  is defined as the average of the space and mark magnitude errors as shown in the following equation:

$$\epsilon_m = \frac{\Delta \mu_s + \Delta \mu_m}{2}$$

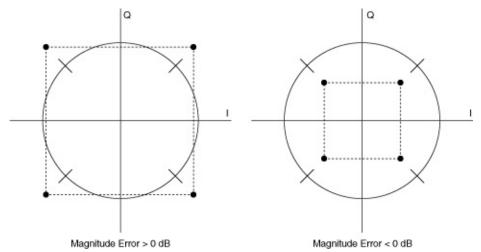
#### where

 $\Delta \mu_{s} = E(\{s_{i} - f_{s}\})$  $\Delta \mu_{m} = E(\{m_{i} - f_{m}\})$ 

for all *i*.

### Magnitude Error, Quadrature Modulation (QM)

In <u>quadrature modulated</u> (QM) systems, *magnitude error* refers to the difference between the ideal magnitude of the I/Q signal and the actual measured magnitude on a symbol-by-symbol basis. Magnitude error in QM systems causes the locus of symbol coordinates to either move away from the origin and expand or toward the origin and shrink. Magnitude error affects both the I and Q components simultaneously and can be either positive or negative, indicating magnitude gain or attenuation, respectively.



#### **Modulation Error Ratio (MER)**

The *modulation error ratio* (MER) is a measure of the signal-to-noise ratio (SNR) in a digitally modulated signal. Like SNR, MER is usually expressed in dB. MER over *N* number of symbols is defined as

$$MER = \frac{\sum_{j=1}^{N} (\tilde{I}_{j}^{2} + \tilde{Q}_{j}^{2})}{\sum_{j=1}^{N} [(I_{j} - \tilde{I}_{j})^{2} + (Q_{j} - \tilde{Q}_{j})^{2}]}$$

where

 $I_j$  is the I component of the *j*-th symbol received

 $\dot{Q}_j$  is the Q component of the *j*-th symbol received

 $\tilde{I}_j$  is the ideal I component of the *j*-th symbol received

 $\tilde{Q}_{j}$  is the ideal Q component of the *j*-th symbol received

#### **Phase Deviation**

In <u>phase modulation</u>, *phase deviation* quantifies the maximum difference between the instantaneous <u>phase angle</u> of the <u>modulated wave</u> and the phase angle of the unmodulated <u>carrier wave</u>.

#### **Phase Error**

In a <u>quadrature modulated</u> (QM) system, *phase error* (shown as  $\theta$  in the equation below) occurs when the measured phase of the received symbols deviates from the ideal phase values. If the symbol sequence is  $\{d_0, d_1, d_2, ..., d_{N-1}\}$ , then the sequence of phase errors is  $\{\theta_0, \theta_1, \theta_2, ..., \theta_{N-1}\}$ . The peak phase error over the *N* symbols is express as

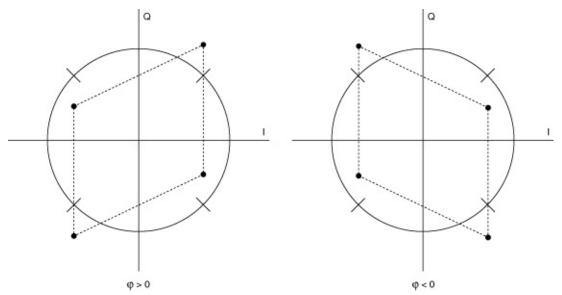
 $\theta_{\text{peak}} = \max(\theta_i)$  i = 0, 1, 2, ..., N-1

and the RMS phase error over N symbols is

$$\theta_{\rm rms} = \int_{N}^{1} \sum_{i=1}^{N} \theta_i^2$$

### **Quadrature Error**

In a <u>quadrature modulated</u> (QM) system, *quadrature error*, also referred to as *quadrature skew*, describes a complex signal impairment such that the I and Q components are not perfectly orthogonal. Quadrature error can be either positive or negative, with the sign indicating the orientation of the error.



# Rho (ρ)

 $\rho$  is a measurement that evaluates modulation quality. It is calculated by finding the correlation between the recovered waveform and an ideal waveform that has been reconstructed from the demodulated bits. That is, the received demodulated waveform is compared to an ideal waveform obtained by remodulating the **output bit stream** of the digital demodulation VI. The  $\rho$  measurement is related to EVM because EVM measures the vector difference between the measured and ideal signals, while  $\rho$  measures the correlation between the two signals. The value of  $\rho$  can range from 0.0 to 1.0. The following table illustrates the ranges for  $\rho$  and degree of correlation between measured and ideal signals.

| <i>ρ</i> = 0.0     | uncorrelated         |
|--------------------|----------------------|
| $0.0 < \rho < 1.0$ | partially correlated |
| $\rho = 1.0$       | perfectly correlated |

# Modulation

The modulation block converts the information signal bit stream into inphase (I) and quadrature-phase (Q) data components. This block typically also involves pulse shaping to minimize intersymbol interference and reduce bandwidth.

Depending on the type of information signal and the particular transmission medium, different modulation techniques are employed. For example, in <u>amplitude modulation (AM)</u>, the information is represented by amplitude variations of the carrier signal.

The Modulation Toolkit supports the following types of analog modulation:

- <u>AM</u>
- <u>FM</u>
- <u>PM</u>

The Modulation Toolkit also supports the following types of digital modulation:

- <u>ASK</u>
- <u>FSK</u>
- <u>MSK</u>
- <u>PAM</u>
- <u>PSK</u>
- <u>QAM</u>
- <u>CPM</u>

# **Modulation Fundamentals**

Expand this book for more information about modulation terminology and signal analysis fundamentals.

#### **Angle Modulation**

*Angle modulation* varies the angle of a <u>carrier wave</u> according to the amplitude of the modulating baseband signal (the <u>message signal</u>). The amplitude of the carrier is kept constant. <u>Phase modulation</u> and <u>frequency modulation</u> are particular types of angle modulation.

Angle modulation can be expressed using the following equation:

$$S_{f_m} = A_c \cos \left[ 2\pi f_c t + \frac{k_f A_m}{f_m} \sin \left( 2\pi f_m t \right) \right]$$

where

 $A_c$  is the carrier amplitude  $f_c$  is the carrier frequency  $k_f$  is the frequency deviation constant in Hz/V  $f_m$  is the frequency of the message signal

# Baseband

The *baseband* is the range in the <u>frequency</u> spectrum occupied by the unmodulated <u>message signal</u>. Both the message signal and the downconverted complex I/Q signal are referred to as baseband signals.

Refer to the <u>NI Developer Zone</u> at ni.com/zone for more information about baseband signals.

#### **Carrier Wave**

The *carrier wave* is a sine wave that is modulated by a <u>message signal</u> prior to transmission. The message signal modifies the carrier wave amplitude, <u>frequency</u>, or <u>phase</u> prior to transmission. During <u>modulation</u>, these characteristics may be varied individually or in combination. The modified carrier signal, also referred to as the <u>modulated wave</u>, is transmitted to a receiver.

The message signal data in the received modulated wave is recovered by removing the carrier signal through <u>demodulation</u>. In advanced communication systems, the carrier may be a moving signal, also known as a spread spectrum. When the characteristics of the carrier signal are deterministic and known by the receiver, virtually any type of carrier signal can be used.

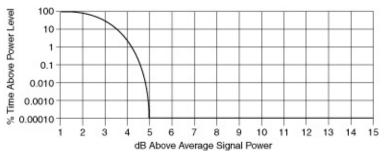
The nominal frequency of the carrier wave is the *carrier frequency*. In <u>frequency modulation</u>, the carrier frequency is the <u>center frequency</u>.

# **Carson's Rule**

*Carson's rule* defines the approximate <u>modulation</u> bandwidth required for a <u>carrier</u> signal that is <u>frequency-modulated</u> by a spectrum of <u>frequencies</u> rather than a single frequency. The Carson bandwidth rule is expressed by the relation CBR = 2 ( $\Delta f + f_m$ ), where CBR is the bandwidth requirement,  $\Delta f$  is the carrier peak deviation frequency, and  $f_m$  is the highest modulating frequency.

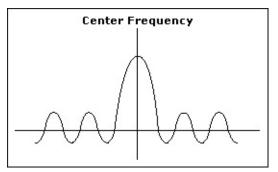
### **CCDF** Measurement

The complementary cumulative distribution function (CCDF) is a statistical characterization of the time-domain waveform that completely describes the power characteristics of a signal. A CCDF graph relates average signal power (X axis) to signal power statistics (Y axis) such that each point on the CCDF curve shows how much time a signal spends at or above a given power level. The power level is expressed in dB relative to the average signal power level.



### **Center Frequency**

The *center frequency* is the middle <u>frequency</u> of the channel bandwidth. In <u>frequency modulation</u>, the center frequency is equal to the *rest frequency*—specifically, the frequency of the unmodulated <u>carrier wave</u>.



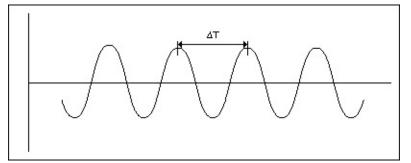
# Demodulation

*Demodulation* describes the recovery, from a <u>modulated wave</u>, of a signal having the same characteristics as the original <u>message signal</u>.

# Frequency

*Frequency* refers to a basic unit of rate measured in events or oscillations per second. Frequency also refers to a number representing a specific point in the electromagnetic spectrum.

The following graph illustrates one period of a sine wave.



Frequency can be represented according to the following equation:

$$f = \frac{1}{\Delta T}$$

where T is the period of one oscillation.

Refer to the <u>NI Developer Zone</u> ni.com/zone for more information about frequency.

# **Intermediate Frequency (IF)**

The intermediate frequency (*IF*) is an intermediate signal that is the product of the RF <u>downconversion</u> process. An RF signal is converted to an IF signal to be digitized, demodulated, displayed, or processed. For example, the NI PXI-5600 downconverter module converts RF signals to IF signals in a band between 5–25 MHz.

# I/Q Data

I/Q data is an alternative method of describing the magnitude and phase data of a signal.

A sinusoidal wave can be written in polar coordinate form as shown in the following equation:

 $f(t) = A\cos(2\pi ft + \varphi)$ 

where

A is the amplitude

 $2\pi f$  is the frequency

 $\phi$  is the phase

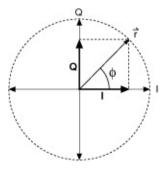
A sinusoidal wave can also be represented in a complex Cartesian coordinate system by its real and complex components such that the inphase (I) component can be written as

 $I(t) = A\cos(\varphi)\cos(2\pi ft)$ 

and the quadrature (Q) component can be written as

 $Q(t) = Asin(\varphi)sin(2\pi ft)$ 

Graphically, I and Q projections of the polar coordinate sinusoidal wave are on the x and y axis, respectively, as illustrated in the following graph.



In the preceding figure, the sinusoidal wave frequency is shown as the rotational rate of the vector 7 around the circle.

The vector magnitude (M) is given by

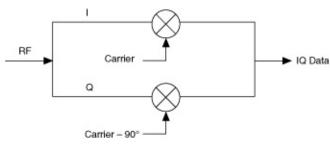
$$M = (I(t)^2 + Q(t)^2)^{1/2}$$

and the vector phase is given by

 $\varphi = \tan^{-1}(Q/I).$ 

While magnitude and phase data seem more intuitive, hardware design concerns make I and Q data the better choice for RF waveforms. I/Q representation provides an effective way to visualize and measure the quality of modulation. The following figure is a generic block diagram of an I/Q demodulator, which takes an RF signal and separates out the I and Q component from that incoming RF signal.

The following figure is a generic block diagram of an I/Q demodulator.



The circles with an 'X' represent mixers. The I/Q modulator is represented here as part of a downconverter module. The incoming <u>message signal</u> splits and one signal is multiplied by an in-phase <u>carrier</u> <u>signal</u> (I) while the other signal is multiplied by a quadrature signal (Q). This multiplication separates the in-phase and quadrature components from the incoming signal.

#### **Message Signal/Information Signal**

The *message signal*, or information signal, contains the data for transmission. The message signal is used to <u>modulate</u> the <u>carrier wave</u> to create the <u>modulated wave</u> for transmission. The message signal data is recovered from the modulated wave by a process of <u>demodulation</u>.

The message signal is often referred to as the **<u>baseband</u>** signal.

#### **Modulated Wave**

The *modulated wave*, or *modulated signal*, refers to the signal for transmission that consists of the <u>carrier wave modulated</u> by the <u>message</u> <u>signal</u>. The message signal is recovered by the receiver through a process of <u>demodulation</u>.

Typically, the incoming wave is an RF signal from a unit under test (UUT).

# Modulation

*Modulation* is a process that alters the characteristics of a <u>carrier wave</u> according to information in the <u>message signal</u> to generate a <u>modulated</u> <u>wave</u> that is transmitted. Modulation Toolkit VIs are capable of analyzing carrier waveforms with <u>amplitude</u>, <u>phase</u>, or <u>frequency</u> modulation.

The unmodulated carrier is represented by the following equation:

 $v(t) = A_c cos(\omega_c t + \theta)$ 

The amplitude-modulated carrier signal is represented by the following equation:

$$v(t) = (m(t) + A_c)\cos(\omega_c t + \theta)$$

The frequency-modulated carrier signal is represented by the following equation:

 $v(t) = A_c cos((m(t) + \omega_c t) + \theta)$ 

The phase-modulated carrier signal is represented by the following equation:

 $v(t) = A_c cos(\omega_c t + \theta + m(t))$ 

where m(t) is the time-varying modulation,  $A_c$  is the amplitude of the carrier wave, and  $\omega_c$  is the frequency of the carrier wave.

# **Modulation Depth**

In <u>amplitude modulation</u>, *modulation depth* refers to the ratio of the unmodulated carrier amplitude to the amplitude deviation for which the <u>modulated carrier wave</u> reaches its minimum value. If this minimum value is zero, the modulation depth is 100%. The *modulation depth* ratio is also referred to as the <u>modulation index</u>.

#### **Modulation Index**

The *modulation index* is the ratio of the frequency deviation of the modulated signal to the message signal bandwidth. For FSK modulation, the modulation index *h* is defined as the ratio of the spacing between consecutive frequencies in the FSK symbol map, to the symbol rate. More precisely,

 $h=2 \times f_d T/(M-1)$ 

where

*M* is the modulation format *T* is the symbol period

 $f_d$  is the peak frequency deviation

For example, in 4-FSK, the spacing between consecutive frequencies equals  $2f_d$  / 3, hence the modulation index equals  $(2f_d \times T)$  / 3.

For more information about amplitude modulation, refer to <u>modulation</u> <u>depth</u>.

# **On-Off Keying (OOK)**

*On-off keying* (OOK) is a modulation scheme that consists of keying a sinusoidal carrier signal on and off with a unipolar binary signal. OOK is equivalent to two-level <u>amplitude-shift keying (ASK)</u>.

|   |      | <br> | 100 |  |
|---|------|------|-----|--|
|   |      |      |     |  |
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|   |      |      |     |  |
|   |      |      |     |  |

**Unipolar Binary Baseband Signal** 



**Resulting OOK Modulated Signal** 

#### Phase

*Phase* refers to periodic changes in waveform magnitude relative to a standard position or instant of starting. For example, the phase of a wave of period T with its starting point at  $t_0$  can be defined in radians:

$$\frac{modulo((t-t_0),T)}{T} \times 2\pi$$

or in degrees:

$$\frac{modulo((t-t_0),T)}{T} \times 360$$

Any vector can be represented either in polar coordinates by  $M \neq 0$ , where *M* is the magnitude and 0 is the <u>phase angle</u>, or in Cartesian coordinates, specifically, an Argand diagram, as (a + jb), where *a* is a real component and *b* is an imaginary component such that  $\tan 0 = (b/a)$ , where 0 is the phase angle, and the magnitude, *M*, is  $(a^{2}+b^{2})^{\frac{1}{2}}$ .

Refer to the <u>NI Developer Zone</u> at ni.com/zone for more information about phase.

## **Phase Angle**

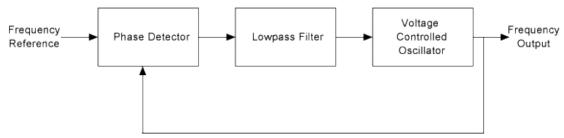
*Phase angle* refers to the angle between a point on a periodic wave and a reference point. The reference point may be a point on another periodic wave.

The angular measurement that defines the relationship between the periodic wave and the reference point is derived from a projection of a rotating vector onto the real axis of an Argand diagram. The phase angle of a point on a wave is the value of the point on the abscissa that corresponds to the point on the wave. The phase angle of a vector may be written as

 $M \neq 0$ , where *M* is the magnitude of the vector and 0 is the phase angle relative to the specified reference.

## **Phase-Locked Loop (PLL)**

A *Phase-locked loop* (PLL) is an electronic circuit that controls an oscillator so that the circuit maintains a constant <u>phase angle</u> relative to a reference signal.



The operation of the above circuit is typical of all PLLs. This circuit is a feedback control system that controls the frequency and phase of a voltage-controlled oscillator (VCO). An input signal is applied to a phase detector and the output of the VCO connects to the other phase detector input. As shown in the previous diagram, the frequencies of both signals are the same. The output of the phase detector develops a voltage proportional to the phase difference between the two input signals. The lowpass filter receives this signal from the phase detector and determines the dynamic characteristics of the PLL. This output signal is the filtered signal that controls the VCO.

## **Modulation Schemes**

Expand this book for more information about supported modulation schemes.

# **Amplitude Modulation (AM)**

*Amplitude modulation* (AM) is a process that varies the amplitude of an RF <u>carrier signal</u> according to the amplitude of the <u>message signal</u>.

The recovery of the message signal is called <u>demodulation</u>. One of the benefits of amplitude modulation systems is the ease with which the <u>baseband</u> message signal can be recovered. Amplitude modulation generates discrete upper and lower sidebands, which are the sum and difference between <u>frequencies</u> of the message signal and the carrier signal.

The following figure illustrates the modulation of a carrier signal, figure **a**, by a message signal, figure **b**. The message signal shape is also referred to as the *amplitude envelope*. The result is the <u>modulated wave</u>, figure **c**.

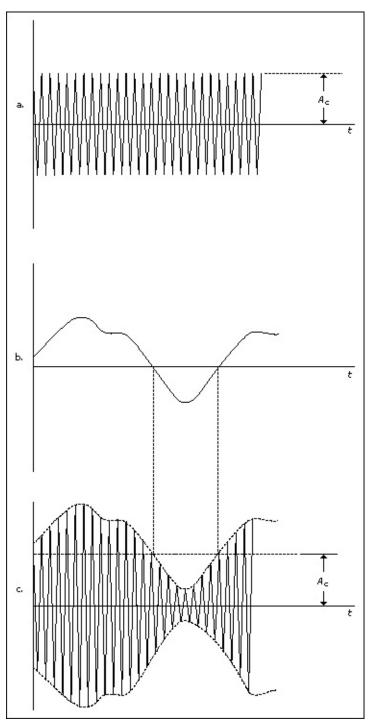


Figure **a** shows a carrier signal with amplitude  $A_c$ . Figure **b** shows the baseband <u>message signal</u>. The transmitted signal in Figure **c** is given by the following equation:

 $v(t) = A_c[1 + m(t)] \times \cos(2\pi f_c t)$ 

where m(t) is the time-varying modulation.

Refer to the <u>NI Developer Zone</u> at ni.com/zone for more information about amplitude modulation.

# **Amplitude-Shift Keying (ASK)**

*Amplitude-shift keying* (ASK) refers to a type of <u>amplitude modulation</u> that assigns bit values to discrete amplitude levels. The carrier signal is then modulated among the members of a set of discrete values to transmit information.

# **Continuous Phase Modulation (CPM)**

Continuous phase modulation (CPM) is a constant-amplitude modulation scheme that can be considered to be a generalization of continuous phase frequency shift-keying (CPFSK) or <u>minimum shift-keying (MSK)</u>. The lack of phase discontinuities reduces high-frequency spectral content, making CPM a highly spectrally efficient scheme.

A form of CPM that can result in significant coding gains is multi-*h* phase coding, where *h* stands for the modulation index. This scheme may be viewed as a generalization of CPFSK schemes because different phase changes result from the transmission of the same symbol in two contiguous symbol intervals. A mathematical representation of the signal during the *i*th interval,  $iT \le t$  (i + 1)T, is expressed by the following formula:

$$x(t,d) = \sqrt{\frac{E_s}{T}} \cos(\omega_c t + d_i \,\omega_i (t - iT) + \varphi_i)$$

where  $E_s$  is the symbol energy

T is the symbol duration

 $\omega_c$  is the carrier frequency in radians/second

d and  $\omega$  are the sequences that represent the M-ary information sequence.

 $d_i\omega_i(t - iT)$  and  $\varphi$  are data phase terms that correspond to the phase associated with the current data symbol and the phase accumulation due to the previous data symbol such that

$$d_i \,\omega_i(t-iT) = \,d_i \,\int_{iT}^T \left(\frac{\pi \boldsymbol{h}_i}{T}\right) g(t_1 - iT) dt_1$$

and

$$\varphi_i = \sum_{k=0}^{i-1} \pi d_k h_k$$
$$\omega_i \left( = \frac{\pi h_i}{T} \right)$$

where is the angular frequency corresponding to the modulation index used during the *i*th baud. The different values of *h* 

can be used between symbol intervals in a round robin fashion.

Refer to the following resources for more information about the algorithms and methods used in CPM:

- Premji, Al-Nasir and Desmond P. Taylor. "Receiver Structures for Multi-h Signaling Formats." *IEEE Transactions on Communications* 35,4 (1987).
- Oerder, Martin and Heinrich Mayer. "Digital Filter and Square Timing Recovery." *IEEE Transactions on Communications* 36, 5 (1988).

## **Frequency Modulation (FM)**

*Frequency <u>modulation</u>* (FM) is a type of <u>angle modulation</u> in which the <u>frequency</u> of a sinusoidal <u>carrier wave</u> deviates from a <u>center frequency</u> by an amount proportional to the instantaneous value of the <u>message</u> <u>signal</u>. In FM, the center frequency is the <u>carrier</u> frequency.

Frequency modulation can be expressed using the general equation for angle modulation.

$$S_{f_m} = A_c \cos \left[ 2\pi f_c t + \frac{k_f A_m}{f_m} \sin \left( 2\pi f_m t \right) \right]$$

where

 $A_c$  is the carrier amplitude

 $f_c$  is the carrier frequency

 $k_f$  is the frequency deviation constant in Hz/V

 $f_m$  is the frequency of the message signal

Refer to the <u>NI Developer Zone</u> at ni.com/zone for more information about frequency modulation.

# **Frequency-Shift Keying (FSK)**

*Frequency-shift keying* (FSK) refers to a type of <u>frequency modulation</u> that assigns bit values to discrete frequency levels. In noncoherent forms of FSK, the instantaneous frequency is shifted between two discrete values termed the *mark* and *space* frequencies. Coherent forms of FSK exist that have no phase discontinuity in the output signal. FSK modulation formats generate modulated waveforms that are strictly real-values, and thus tend not to share common features with <u>quadrature</u> modulation (QM) schemes.

# Minimum Shift-Keying (MSK)

*Minimum shift keying* (MSK) modulation is a subtype of **FSK** modulation. MSK uses a half-cycle sinusoidal pulse, making the phase change linear and keeping side lobes low to control adjacent-channel interference.

### **Phase Modulation (PM)**

*Phase modulation* (PM) is a type of <u>angle modulation</u> in which the <u>phase</u> <u>angle</u> of a <u>carrier wave</u> is made to deviate from its reference value by an amount proportional to the instantaneous value of the modulating <u>message signal</u>. The resulting phase-<u>modulated wave</u> is transmitted.

Phase modulation can be expressed using the following general equation for angle modulation:

$$S_{f_m} = A_c \cos \left[ 2\pi f_c t + \frac{k_f A_m}{f_m} \sin \left( 2\pi f_m t \right) \right]$$

where

 $A_c$  is the carrier amplitude

 $f_c$  is the carrier frequency

 $k_f$  is the frequency deviation constant in Hz/V

 $f_m$  is the frequency of the message signal

Refer to the <u>NI Developer Zone</u> at ni.com/zone for more information about phase modulation.

## **Phase-Shift Keying (PSK)**

*Phase-shift keying* (PSK) in a digital transmission refers to a type of angle modulation in which the phase of the carrier is discretely varied to represent data being transmitted—either in relation to a reference phase or to the phase of the immediately preceding signal element.

For example, when encoding bits, the phase shift could be 0° for encoding a 0 and 180° for encoding a 1, or the phase shift could be -90° for 0 and +90° for a 1, thus making the representations for 0 and 1 a total of 180° apart. In PSK systems designed so that the carrier can assume only two different phase angles, each change of phase carries one bit of information, that is, the bit rate equals the modulation rate. If the number of recognizable phase angles is increased to four, then 2 bits of information can be encoded into each signal element; likewise, eight phase angles can encode 3 bits in each signal element.

# **Quadrature Modulation (QM)**

*Quadrature modulation* (QM) refers to any <u>modulation</u> scheme that uses two <u>carrier</u> waves out of <u>phase</u> by 90° that are modulated by separate <u>information signals</u>.

The QM formats available in this toolkit are <u>phase-shift-keying (PSK)</u>, <u>quadrature-amplitude modulation (QAM)</u>, and <u>minimum-shift keying (MSK)</u>.

## Quadrature Modulated (QM) Waveform, Ideal

In a <u>quadrature modulates</u> (QM) system, the *QM ideal waveform* is the sum of the <u>I</u> and <u>Q</u> component of a signal as follows:

$$s(t) = s_I(t) + s_O(t)$$

To obtain the QM waveform, the baseband components are modulated orthogonally as

$$s_{I}(t) = i(t)\cos(\omega_{c}t)$$
  
$$s_{Q}(t) = q(t)\sin(\omega_{c}t)$$

thus

 $s(t) = i(t)\cos(\omega_c t) + q(t)\sin(\omega_c t)$ 

where i(t) and q(t) are the baseband I and Q waveforms, respectively.

### **Quadrature Modulated (QM) Waveform, Practical**

In a <u>quadrature modulated</u> (QM) system, the *QM practical waveform* differs from the <u>QM ideal waveform</u>. A generalized adjusted QM waveform can be expressed as

 $s(t) = (\alpha_I i(t) + \Delta_I) \cos(\omega_c t) + (\alpha_O q(t) + \Delta_O) \sin(\omega_c t + \varphi)$ 

where

 $\alpha_{I}/\alpha_{Q}$  is the I/Q gain imbalance

 $\Delta_l$  is the in-phase DC offset

 $\Delta_O$  is the quadrature DC offset

 $\phi$  is the quadrature error

# **Quadrature-Amplitude Modulation (QAM)**

*Quadrature-amplitude modulation* (QAM) is a form of <u>quadrature</u> <u>modulation</u> in which the two carriers are both amplitude-modulated.

# **Quadrature-Phase Shift-Keying (QPSK)**

*Quadrature-phase shift keying* (QPSK) is a form of <u>phase-shift keying</u> in which four different phase angles are used. In QPSK, the four angles are usually separated by 90° spacing.

## Upconversion

The <u>baseband</u> modulated signal undergoes analog upconversion to frequency-translate the signal to the RF frequency at which the signal is transmitted.

Use the <u>MT Upconvert Baseband</u> VI to upconvert waveforms.

# Visualization

Visualization tools, such as constellation plots and eye diagrams, are used to visualize communications measurements.

- Modulation Toolkit supports the following types of digital visualization:
  - <u>3D Eye Diagram</u>
  - 2D Eye Diagram
  - Constellation Plot
  - Trellis Diagram
  - <u>XY graph</u>

# **Operating System Support**

For information about the supported operating system (OS) for the NI Modulation Toolkit, refer to the Modulation Readme.

# **Programming Examples**

The Modulation Toolkit includes several examples for LabVIEW. These examples serve as interactive tools, programming models, and building blocks in your own applications.

With LabVIEW running, select **Help»Find Examples** to launch the LabVIEW Example Finder. The Example Finder offers two ways to access all installed LabVIEW example VIs and their descriptions:

- Click the **Browse** tab to locate modulation VI examples by task at **Toolkits and Modules**»**Modulation** or by directory structure at **Modulation**.
- Click the **Search** tab to search all installed examples by keyword. Enter the keyword FM, for instance, to locate a examples demonstrating FM modulation and demodulation.

For the location of the installed modulation VI example files, refer to the <u>NI Modulation Toolkit for LabVIEW Readme</u>.

#### **Examples Available Online**

Modulation Toolkit VI examples are also available online at the <u>NI</u> <u>Developer Zone</u> or at ni.com/examples. Refer to the NI Developer Zone at ni.com/zone for more information about integrating the Modulation Toolkit with the RF signal analyzer and RF signal generator.

### Considerations for Using the LabVIEW Real-Time Module

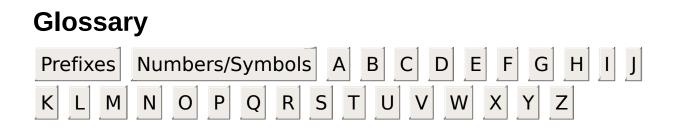
To develop a Modulation Toolkit for LabVIEW application in the LabVIEW Real-Time Module, follow the same steps used for developing any application in the LabVIEW Real-Time Module, with the addition of using the Modulation Toolkit for LabVIEW VIs.

#### 🕅 Note

- Applications running Modulation Toolkit in the LabVIEW Real-Time Module on an RT target may be compromised and/or slow at 64 MB. Using Modulation Toolkit VIs may introduce high jitter into the entire system. NI does not recommend using these VIs in applications with deterministic requirements.
- The Modulation Toolkit supports the LabVIEW Real-Time Module 8.5 and later.
- The Modulation Toolkit is supported only on PXI Real-Time systems.

#### **Related Documentation**

- For configuration instructions for remote systems, refer to the MAX Remote Systems Help in Measurement & Automation Explorer (MAX) by selecting Help»Help Topics»Remote Systems in MAX.
- For more information about the LabVIEW Real-Time Module, refer to the *LabVIEW Real-Time Module User Manual* at <u>ni.com/manuals</u>.
- For additional troubleshooting and support information, refer to the LabVIEW Real-Time Support main page at <u>ni.com/support/labview/real-time</u>.



#### **Prefixes**

| Symbol | Prefix | Value            |
|--------|--------|------------------|
| n      | nano   | 10 <sup>-9</sup> |
| μ      | micro  | 10 -6            |
| m      | milli  | 10 -3            |
| k      | kilo   | 10 <sup>3</sup>  |
| М      | mega   | 10 6             |
| G      | giga   | 10 <sup>9</sup>  |

#### Numbers/Symbols

|    | . –        |                          |
|----|------------|--------------------------|
| nV | nanovolts  | 10 <sup>-9</sup> volts   |
| μV | microvolts | 10 <sup>-6</sup> volts   |
| μΩ | microohms  | 10 <sup>-6</sup> ohms    |
| mΩ | milliohms  | 10 <sup>-3</sup> ohms    |
| MΩ | megaohms   | 10 <sup>6</sup> ohms     |
| nA | nanoamps   | 10 <sup>-9</sup> amperes |
| μA | microamps  | 10 <sup>-6</sup> amperes |
| mΑ | milliamps  | 10 <sup>-3</sup> amperes |
|    |            |                          |

#### Α

| amplitude<br>droop                          | Measured in dB, is a measure of the amount that the signal power falls from the start of a specified measurement window $(d_i)$ to the end of that window $(d_f)$ .   |
|---|---|
| amplitude<br>modulation<br>(AM)             | A process that varies the amplitude of an radio frequency (RF) <u>carrier signal</u> according to the amplitude of the message signal.  |
| amplitude-<br>shift<br>keying<br>(ASK)      | Refers to a type of <u>amplitude modulation</u> which assigns bit<br>values to discrete amplitude levels. The carrier signal is<br>then modulated among the members of a set of discrete<br>values to transmit information. |
| analog-to-<br>digital<br>converter<br>(ADC) | A hardware component that converts analog voltages to<br>digitized values. An ADC can convert an analog signal to a<br>digital signal representing equivalent information.  |

#### В

bit The ratio of erroneous bits to total bits transmitted, received, or

error processed over some stipulated period. Transmission BER

rate expresses the number of erroneous bits received divided by the

(BER) total number of bits transmitted. Information BER expresses the number of erroneous decoded (corrected) bits divided by the total number of decoded (corrected) bits.

burst For burst signals, burst timing refers to the location of the burst,

timing obtained by its correlation against an ideal power curve. In addition, an upper and lower mask are used for testing whether the burst signal satisfies mask specifications. The following figure shows upper mask, lower mask, and ideal power curve.

#### С

| Carson's<br>Rule    | Defines the approximate modulation bandwidth required<br>for a <u>carrier</u> signal that is <u>frequency-modulated</u> by a<br>spectrum of <u>frequencies</u> rather than a single frequency.<br>The Carson bandwidth rule is expressed by the relation<br>CBR = $2(\Delta f + f_m)$ where CBR is the bandwidth<br>requirement, $\Delta f$ is the carrier peak deviation frequency,<br>and $f_m$ is the highest modulating frequency. |
|---------------------|--|
| CCDF<br>measurement | The complementary cumulative distribution function (CCDF) is a statistical characterization of the time-<br>domain waveform that completely describes the power characteristics of a signal.   |
| center<br>frequency | The middle <u>frequency</u> of the channel bandwidth. In <u>frequency modulation</u> , the center frequency is equal to the <i>rest frequency</i> —specifically, the frequency of the unmodulated <u>carrier wave</u> .  |
| code word           | The generated coded bits/numbers from a channel coding system.   |
| complex<br>envelope | A complex representation of the <u>baseband</u> <u>modulated</u><br>signal.  |
| component           | The real and imaginary parts of a complex number are referred to as <i>components</i> . Modulation Toolkit VIs can use complex components to describe signal properties.   |
|                     | For example, you can represent a two-dimensional vector<br>of length <i>S</i> by its components $S = A + iB$ , where <i>A</i> and <i>B</i><br>are the vector x- and y-components. The real part of the<br>vector corresponds to the x-component ( <i>A</i> ), while the<br>imaginary part corresponds to the y-component ( <i>B</i> ).   |

#### D

data word The incoming message bits to a channel coding system.

- DC offset A complex signal impairment that shifts the locus of ideal symbol coordinates off-center in the I/Q plane. A DC offset can be added to the baseband I component, the Q component, or both. The DC offset can be either positive or negative, with the sign indicating direction of the shift. DC offset is expressed as a percentage of full scale, where "full scale" (fs) is the amplitude of the baseband QM waveform.
- depuncture The process of inserting erasure values into the input data stream prior to its input to the decoder. If the input data is real-valued BPSK modulated data (as in the case of unquantized symbol decisions from a demodulator or equalizer), the erasure values equal 0. If the input data stream consists of quantized integers coming from an A/D converter (ADC) at the output of a <u>demodulator</u>, the erasure values correspond to the integer representation that is half the maximum output sample value generated by the analog-to-digital conversion process.
- deviation error For an M-FSK system, the average deviation error is defined as the average magnitude of the spread of the FSK demodulated symbol spaced waveform around the ideal symbol (frequency) locations.

Mathematically, the deviation error is defined as:

$$f_{dev} = \frac{1}{M} \left| \sum_{i=0}^{M-1} \left( f_{ideal,i} - \left\langle f_{actual,i} \right\rangle \right) \right|$$

where *M* is the FSK modulation format,  $f_{ideal,i}$  is the ideal symbol location at the FSK frequency corresponding to location *i*, and  $< f_{actual,i} >$  is the mean value of the demodulated symbols at location *i*.

digital-to-<br/>analogA hardware component that converts digital values to<br/>analog voltages. Thus a DAC can convert a digital

converter signal to an analog signal representing equivalent information.

direct sequence spread spectrum (DSSS) A process by which data is transmitted using a higher bandwidth signal that is demanded by the data rate. Using DSSS allows multiple channels to occupy the same bandwidth, mitigating interference from other users at the expense of bandwidth expansion.

DSSS is accomplished by spreading each bit of signal data is spread at the transmitter into *L* chips, using a pseudorandom *L*-chip spreading code called a *code word*. The length *L* of the pseudorandom spreading code is also known as the *bandwidth expansion factor* because the chips are transmitted at a rate equal to  $L \times bit$  rate of the data. The spreading code appears random to all receivers except the intended one, which uses the knowledge of the spreading code to demodulate and recover the transmitted information. Thus multiple channels can occupy the same portion of the frequency spectrum by using code words that have little or no correlation with one another, and little or no autocorrelation for any shift other than zero.

Mathematically, a DSSS signal is described by:

$$y(t) = \sum_{n=-\infty}^{\infty} \sum_{m=0}^{L-1} a_n c_m g \left( \tau - n\tau - m\tau_c \right)$$

where

y(t) is the transmitted DSSS signal g(t) is the pulse-shaping signal of duration  $T_c$   $\{a_i\}$  is the i<sup>th</sup> information bearing symbol  $\{c_k\}$  is the k<sup>th</sup> element of the *L*-long pseudorandom spreading code (also known as the chip sequence)  $T_c$  is the chip period, and  $T=L \times T_c$  is the symbol period

downconverter A signal conditioning device that converts a specific band of high-frequency (RF) signals to more manageable <u>intermediate frequencies</u> (IF) that can be digitized.

F

frequency Refers to a basic unit of rate measured in events or oscillations per second. Frequency also refers to a number representing a specific point in the electromagnetic spectrum.

#### I

information Contains the data for transmission. The information signal is signal used to modulate the carrier wave to create the modulated wave for transmission. The information signal data is recovered from the modulated wave by a process of demodulation.

The information signal is often referred to as the <u>baseband</u> signal or *message signal*.

interleaver A device that ensures the symbols from several different code words are well separated during transmission over a single path, so that the symbols from any given code word are clearly received in time-division sequence. Interleavers are used in conjunction with error-correcting codes to counteract the effects of burst errors.

#### Μ

message Contains the data for transmission. The message signal is signal used to <u>modulate</u> the <u>carrier wave</u> to create the <u>modulated</u> <u>wave</u> for transmission. The message signal data is recovered from the modulated wave by a process of <u>demodulation</u>.

The message signal is often referred to as the <u>baseband</u> signal or *information signal*.

mixer A nonlinear analog circuit that multiplies two signals. Mixers are typically used to shift signal <u>frequencies</u>. A mixer receives two sinusoidal input signals at different frequencies and returns a signal with components at frequencies equal to the sum and difference of the two original input frequencies. Nonlinear mixers are used when performing <u>amplitude</u> <u>modulation</u> of RF <u>carrier</u> signals.

#### Ν

noise The ratio of the actual output noise to the noise that would remain figure if the instrument did not contribute its own thermal noise. In

(NF) heterodyne systems, output noise power includes spurious contributions from image-frequency transformation. However, the portion attributable to thermal noise in the input termination includes only what appears in the output due to the principal frequency transformation of the system, and it excludes what appears via the image frequency transformation.

#### 0

offset A variant of phase-shift keying modulation using 4 quadrature different values of the phase to transmit the signal. This scheme is sometimes referred to as staggered quadrature phase-shift keying (SQPSK). (OQPSK)

#### Ρ

phase- An electronic circuit that controls an oscillator so that the locked circuit maintains a constant <u>phase angle</u> relative to a reference signal.
(PLL)

puncture The process of artificially increasing the code rate of the data stream, generated from a block or convolutional encoder, by selectively deleting certain elements in the data stream.

#### R

radio refers to the radio frequency range of the electromagnetic frequency spectrum. RF is often used to describe a range of sub-(RF) infrared <u>frequencies</u> from the tens of MHz to several GHz.

RF signal refers to a family of PXI and PXI Express (PXIe) devices that analyzer include the NI PXI-5660, the NI PXI-5661, and the NI 5663 (RFSA) RF vector signal analyzers.

| Device<br>Name   | Device Components  |  |
|------------------|--|--|
| NI PXI-<br>5660  | NI PXI-5600 RF downconverter module and an NI PXI-5620 IF digitizer module   |  |
| NI PXI-<br>5661  | NI PXI-5600 RF downconverter module and an NI PXI-5142 IF digitizer module   |  |
| NI PXIe-<br>5663 | NI PXIe-5601 RF downconverter module, an<br>NI PXIe-5622 IF digitizer module, and an NI 5652<br>local oscillator (LO) source |  |

The NI 5660 uses the ni5660 Vis in LabVIEW and the NI-TUNER and NI-SCOPE instrument drivers in C, C++, and LabWindows™/CVI™. The NI 5661 and NI 5663 use the NI-RFSA driver for controlling the RF downconverter module, the RF digitizer module, and an LO source (NI 5663 only). All NI RF signal analyzers include the NI Spectral Measurements Toolkit for performing frequency-domain analysis, and modulation VIs for performing analog modulation and demodulation measurements.

#### S

sample The sample rate is the rate at which a device acquires an analog signal, expressed in samples per second (S/s). The sample rate is usually the clock speed of the analog-to-digital converter (ADC).

signal-to- The ratio of the desired signal amplitude to the noise signal

noise ratio amplitude at a given point in time. SNR is expressed as 20 (SNR) times the logarithm of the amplitude ratio, or 10 times the logarithm of the power ratio. SNR is usually expressed in dB and in terms of peak values for impulse noise and root mean square (RMS) values for random noise. In defining or specifying the SNR, specify the signal and noise characterizations, for example, peak-signal-to-peak-noise ratio to avoid ambiguity.

signal-to- A measurement of the effect of quantization errors quantized- introduced by analog-to-digital conversion at the analog-tonoise ratio digital converter (ADC). Exceeding the SQNR of your (SQNR) instrument clips the signal.

- spectral A measure of total signal power in a specified spectral bandwidth divided by the bandwidth, expressed in watts per hertz (W/Hz).
- symbol Expresses the number of symbols transmitted per second (symbols/s). To convert symbol rate into bit rate, which expresses the number of bits transferred per second, multiply the symbol rate by the number of bits per symbol used in the digital modulation scheme of interest. Symbol rate is also known as *baud rate*.

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| Slovenia       | 386 3 425 42 00         |

| South Africa              | 27 0 11 805 8197    |
|---------------------------|---------------------|
| Spain                     | 34 91 640 0085      |
| Sweden                    | 46 (0) 8 587 895 00 |
| Switzerland               | 41 56 2005151       |
| Taiwan                    | 886 02 2377 2222    |
| Thailand                  | 662 278 6777        |
| Turkey                    | 90 212 279 3031     |
| United Kingdom            | 44 (0) 1635 523545  |
| United States (Corporate) | 512 683 0100        |