Instructions

Tektronix

TDS Family
Option 2F Advanced DSP Math

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Welcome

This instruction manual describes Option 2F, the Advanced DSP Math option. Included in this manual are the following subsections:

- Product Description (follows this introduction)
- Fast Fourier Transforms
- Waveform Differentiation
- Waveform Integration

Related Manuals

This manual only documents Option 2F. The following documents cover all other aspects related to the use or service of the oscilloscope. If ordering any of these manuals, order by the manual title and the model of your oscilloscope.

- The *TDS User Manual* guides the user in operation of this oscilloscope and describes its features. It also contains a tutorial, a specification, and other useful information related to your oscilloscope.
- The *TDS Series Programmer Manual* describes how to use a computer to control the oscilloscope through the GPIB interface.
- The *TDS Reference* gives you a quick overview of how to operate your oscilloscope.
- The *TDS Performance Verification* tells how to verify the performance of the oscilloscope.
- The *TDS Service Manual* provides information for maintaining and servicing your oscilloscope.

Conventions

In this manual, you will find various procedures that contain steps of instructions for you to perform. To keep those instructions clear and consistent, this manual uses the following conventions:

- Names of front panel controls and menu labels appear in boldface print.
- Names also appear in the same case (initial capitals, all uppercase, etc.) in the manual as is used on the oscilloscope front panel and menus. Front panel names are all upper case letters, for example, **VERTICAL MENU**, **CH 1**, etc.
- Instruction steps are numbered. The number is omitted if there is only one step.

- When steps require that you make a sequence of selections using front panel controls and menu buttons, an arrow (➞) marks each transition between a front panel button and a menu, or between menus. Also, whether a name is a main menu or side menu item is clearly indicated: Press **VERTICAL MENU** ➞ **Offset** (main) ➞ **Set to 0 V** (side) ➞ **Position** (main) ➞ **Set to 0 divs** (side).

  Using the convention just described results in instructions that are graphically intuitive and simplifies procedures. For example, the instruction just given replaces these five steps:

  1. Press the front panel button **VERTICAL MENU**.
  2. Press the main menu button **Offset**.
  3. Press the side-menu button **Set to 0 V**.
  4. Press the main menu button **Position**
  5. Press the side menu button **Set to 0 divs**

- Sometimes you may have to make a selection from a popup menu: Press **SHIFT UTILITY** ➞ **SYSTEM** (popup). In this example, you repeatedly press the main menu button **SYSTEM** until **I/O** (for example) is highlighted in the pop-up menu.

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Safety

Please take a moment to review these safety precautions. They are provided for your protection and to prevent damage to the digitizing oscilloscope. This safety information applies to all operators and service personnel.

Symbols and Terms

These two terms appear in manuals:

- CAUTION statements identify conditions or practices that could result in damage to the equipment or other property.
- WARNING statements identify conditions or practices that could result in personal injury or loss of life.

These two terms appear on equipment:

- *CAUTION* indicates a personal injury hazard not immediately accessible as one reads the marking, or a hazard to property including the equipment itself.
- *DANGER* indicates a personal injury hazard immediately accessible as one reads the marking.

This symbol appears in manuals:

Static-Sensitive Devices

These symbols appear on equipment:

DANGER
High Voltage

Protective
ground (earth)
terminal

ATTENTION
Refer to
manual

Specific Precautions

Observe all of these precautions to ensure your personal safety and to prevent damage to either the digitizing oscilloscope or equipment connected to it.

Power Source

The digitizing oscilloscope is intended to operate from a power source that will not apply more than 250 V_{RMS} between the supply conductors or between either supply conductor and ground. A protective ground connection, through the grounding conductor in the power cord, is essential for safe system operation.

Grounding the Digitizing Oscilloscope

The digitizing oscilloscope is grounded through the power cord. To avoid electric shock, plug the power cord into a properly wired receptacle where earth ground has been verified by a qualified service person. Do this before making connections to the input or output terminals of the digitizing oscilloscope.

Without the protective ground connection, all parts of the digitizing oscilloscope are potential shock hazards. This includes knobs and controls that may appear to be insulators.

Use the Proper Power Cord

Use only the power cord and connector specified for your product. Use only a power cord that is in good condition.

Use the Proper Fuse

To avoid fire hazard, use only the fuse specified in the parts list for your product, matched by type, voltage rating, and current rating.

Do Not Remove Covers or Panels

To avoid personal injury, do not operate the digitizing oscilloscope without the panels or covers.

Electric Overload

Never apply to a connector on the digitizing oscilloscope a voltage that is outside the range specified for that connector.

Do Not Operate in Explosive Atmospheres

The digitizing oscilloscope provides no explosion protection from static discharges or arcing components. Do not operate the digitizing oscilloscope in an atmosphere of explosive gases.

Product Description

Option 2F, Advanced DSP Math, is an option to the TDS Family Digitizing Oscilloscopes. It can be ordered only at the time the oscilloscope is purchased.

TDS oscilloscopes use a proprietary Digital Signal Processor (DSP) to convert normally acquired waveforms into simple math waveforms (inverts a waveform, adds, subtracts, multiplies two waveforms). Option 2F adds three new complex math waveforms to the oscilloscope:

FFT (Fast Fourier Transform) Math—transforms a displayed waveform from the time domain to the frequency domain by applying the Fast Fourier Transform. FFT Math features the following capabilities:

- Displays the magnitude of the various frequencies the source waveform contains or, optionally, the phase angle of those frequencies
- Measures magnitude in linear V_{RMS} or dB with respect to 1 V_{RMS}; measures phase in degrees or radians
- Provides phase suppression to reduce the phase angle to zero for frequencies with magnitudes below a user-specified threshold
- Transforms source waveforms with record lengths of 500, 1,000, 5,000, 15,000, 30,000, 50,000, and 60,000 points. (Your model oscilloscope will not have all of these these record lengths; consult your User manual to determine which are available with your TDS model.)
- Operates on source waveforms displayed in any channel or reference memory (the source cannot be another math waveform)
- Provides the following FFT windows to optimize frequency resolution and magnitude measurement accuracy: rectangular, Hamming, Hanning, and Blackman-Harris
- Executes an FFT in as little as 64.2 milliseconds (1,000 point FFT)
- Updates the display up to ten times per second (1,000 point FFT)
- Corrects FFT DC error automatically
- Allows cursor and automatic measurements of FFT math waveforms

Derivative Math—differentiates, with respect to time, a waveform displayed from a channel or from a reference memory. Derivative math waveforms measure the instantaneous rate of change of a waveform.

Integral Math—integrates over time a waveform displayed from a channel or from a reference memory. Integral math waveforms display the total area under a waveform with respect to ground.

Fast Fourier Transforms

Fast Fourier Transforms

Using the Fast Fourier Transform (FFT), you can transform a waveform from a display of its amplitude against time to one that plots the amplitudes of the various discrete frequencies the waveform contains. Further, you can also display the phase shifts of those frequencies. Use FFT math waveforms in the following applications:

- Testing impulse response of filters and systems
- Measuring harmonic content and distortion in systems
- Characterizing the frequency content of DC power supplies
- Analyzing vibration
- Analyzing harmonics in 50 and 60 cycle lines
- Identifying noise sources in digital logic circuits

Description

The FFT computes and displays the frequency content of a waveform you acquire as an FFT math waveform. This *frequency domain* waveform is based on the following equation:

$$X(k) = \frac{1}{N} \sum_{n = \frac{-N}{2}}^{\frac{N}{2} - 1} x(n) e^{-\frac{j2\pi nk}{N}} \quad for: k = 0 \ to \ N - 1$$

Where: *x(n) is a point in the time domain record data array*

X(k) is a point in the frequency domain record data array

n is the index to the time domain data array

k is the index to the frequency domain data array

N is the FFT length

j is the square root of –1

The resulting waveform is a display of the magnitude or phase angle of the various frequencies the waveform contains with respect to those frequencies. For example, Figure 1-1 shows the non-transformed impulse response of a system in channel 2 at the top of the screen. The FFT-transformed magnitude and phase appear in the two math waveforms below the impulse. The horizontal scale for FFT math waveforms is always expressed in frequency/division with the beginning (left-most point) of the waveform representing zero frequency (DC).

The FFT waveform is based on digital signal processing (DSP) of data, which allows more versatility in measuring the frequency content of waveforms. For example, DSP allows the oscilloscope to compute FFTs of source waveforms that must be acquired based on a single trigger, making it useful for measuring the frequency content of single events. (The TDS 800 model oscilloscopes must have repetitive triggers; therefore, they cannot compute FFTs of single events.) DSP also allows the phase as well as the magnitude to be displayed.

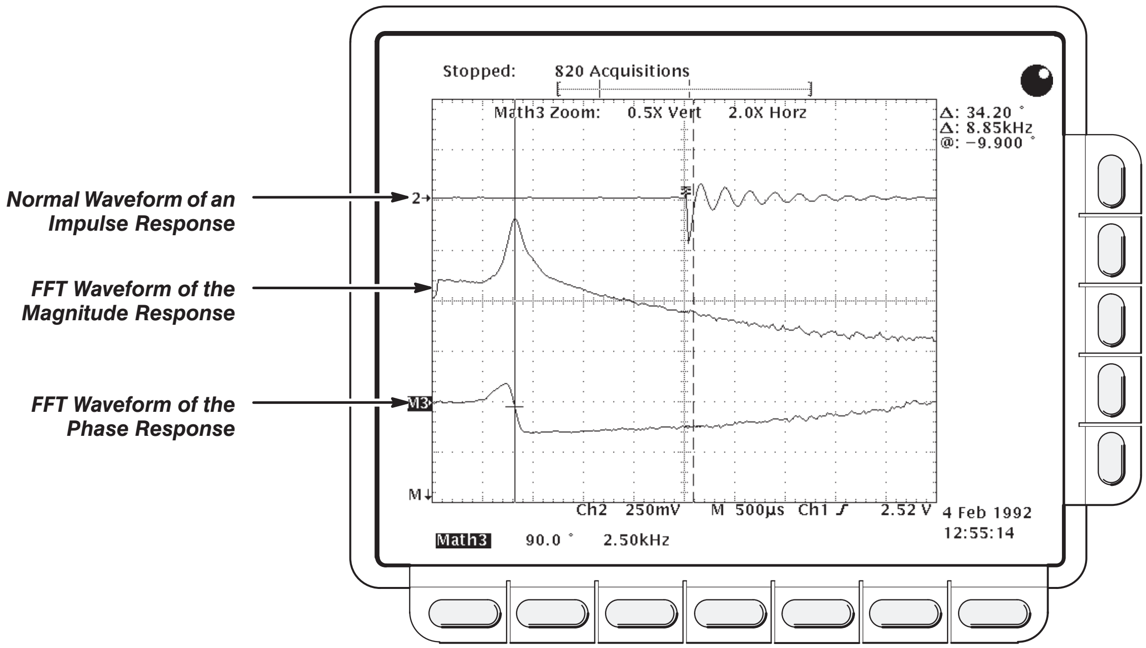


Figure 1-1: System Response to an Impulse

Operation

To obtain an FFT of your waveform, do these basic tasks:

- Acquire and display it normally (that is, in the time domain) in your choice of input channels.
- Transform it to the frequency domain using the math waveform menu.
- Use cursors or automated measurements to measure its parameters.

Use the following procedure to perform these tasks.

Displaying an FFT

1. Connect the waveform to the desired channel input and select that channel.

2. Adjust the vertical and horizontal scales and trigger the display (or press **AUTOSET**).

   The topic *Offset, Position, and Scale,* on page 1-9, provides in depth information about optimizing your setup for FFT displays.

3. Press **MORE** to access the menu for turning on math waveforms.

4. Select a math waveform. Your choices are **Math1**, **Math2**, and **Math3** (main).

5. Press **Change Math Definition** (side) → **FFT** (main). See Figure 1-2.

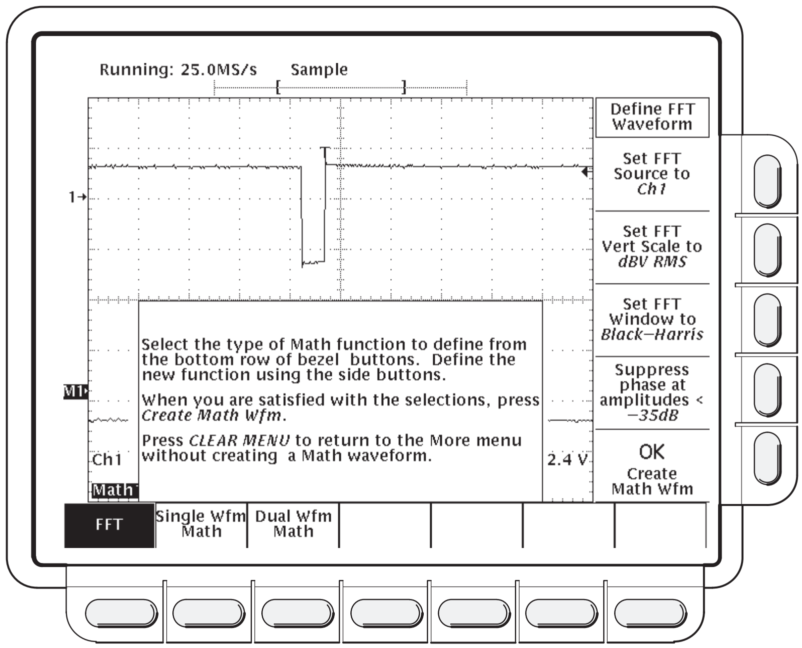


Figure 1-2: Define FFT Waveform Menu

6. Press **Set FFT Source to** (side) repeatedly (or use the general purpose knob) until the channel source selected in step 1 appears in the menu label.

7. Press **Set FFT Vert Scale to** (side) repeatedly to choose from the following vertical scale types:

   - **dBV RMS**—Magnitude is displayed using log scale, expressed in dB relative to 1 V_{RMS} where 0 dB =1 V_{RMS}.
   - **Linear RMS**—Magnitude is displayed using voltage as the scale.
   - **Phase (deg)**—Phase is displayed using degrees as the scale, where degrees wrap from −180° to +180°.

- **Phase (rad)**—Phase is displayed using radians as the scale, where radians wrap from $-\pi$ to $+\pi$.

The topic *Considerations for Phase Displays,* on page 1-12, provides in depth information on setup for phase displays.

8. Press **Set FFT Window to** (side) repeatedly to choose from the following window types:

   - **Rectangular**—Best type for resolving frequencies that are very close to the same value but worst for accurately measuring the amplitude of those frequencies. Best type for measuring the frequency spectrum of non-repetitive signals and measuring frequency components near DC.
   - **Hamming**—Very good window for resolving frequencies that are very close to the same value with somewhat improved amplitude accuracy over the rectangular window.
   - **Hanning** —Very good window for measuring amplitude accuracy but degraded for resolving frequencies.
   - **Blackman-Harris**—Best window for measuring the amplitude of frequencies but worst at resolving frequencies.

   The topic *Selecting the Window,* on page 1-14, provides in depth information on choosing the right window for your application.

9. If you did not select **Phase (deg)** or **Phase (rad)** in step 7, skip to step 12. Phase suppression is only used to reduce noise in phase FFTs.
10. If you need to reduce the effect of noise in your phase FFT, press **Suppress phase at amplitudes** (side).
11. Use the general purpose knob (or the keypad if your oscilloscope is so equipped) to adjust the phase suppression level. FFT magnitudes below this level will have their phase set to zero.

    The topic *Adjust Phase Suppression,* on page 1-13, provides additional information on phase suppression.

12. Press **OK Create Math Wfm** (side) to display the FFT of the waveform you input in step 1.

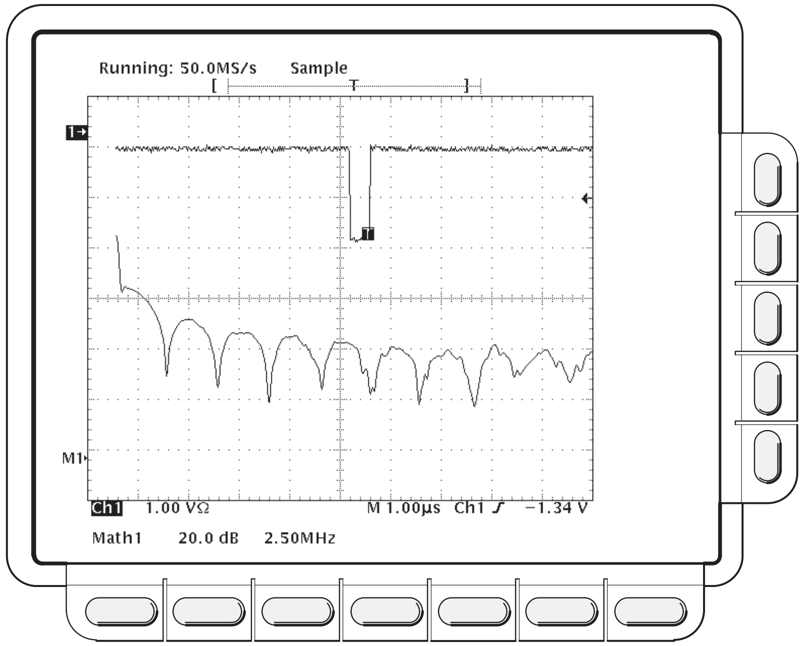


Figure 1-3: FFT Math Waveform in Math1

Cursor Measurements of an FFT

Once you have displayed an FFT math waveform, use cursors to measure its frequency amplitude or phase angle.

1. Be sure **MORE** is selected in the channel selection buttons and that the FFT math waveform is selected in the **More** main menu.
2. Press **CURSOR** → **Mode** (main) → **Independent** (side) → **Function** (main) → **H Bars** (side).
3. Use the general purpose knob to align the selected cursor (solid line) to the top (or to any amplitude on the waveform you choose).
4. Press **TOGGLE** to select the other cursor. Use the general purpose knob to align the selected cursor to the bottom (or to any amplitude on the waveform you choose).
5. Read the amplitude between the two cursors from the **Δ:** readout. Read the amplitude of the selected cursor relative to either 1 V_{RMS} (0 dB), ground (0 volts), or the zero phase level (0 degrees or 0 radians) from the **@:** readout. (The waveform reference indicator at the left side of the graticule indicates the level where phase is zero for phase FFTs.)

   Figure 1-4 shows the cursor measurement of a frequency magnitude on an FFT. The **@:** readout reads 0 dB because it is aligned with the 1 V_{RMS} level. The **Δ:** readout reads 24.4 dB indicating the magnitude of the frequency it is measuring is −24.4 dB relative to 1 V_{RMS}. The source waveform is turned off in the display.

The cursor units will be in dB or volts for FFTs measuring magnitude and in degrees or radians for those FFTs measuring phase. The cursor unit depends on the selection made for **Set FFT Vert Scale to** (side). See step 7 on page 1-3 for more information.

6. Press **V Bars** (side). Use the general purpose knob to align one of the two vertical cursors to a point of interest along the horizontal axis of the waveform.
7. Press **TOGGLE** to select the alternate cursor.
8. Align the alternate cursor to another point of interest on the math waveform.
9. Read the frequency difference between the cursors from the **Δ:** readout. Read the frequency of the selected cursor relative to the zero frequency point from the **@:** readout.

   The cursor units will always be in Hz, regardless of the setting in the **Time Units** side menu. The first point of the FFT record is the zero frequency point for the **@:** readout.

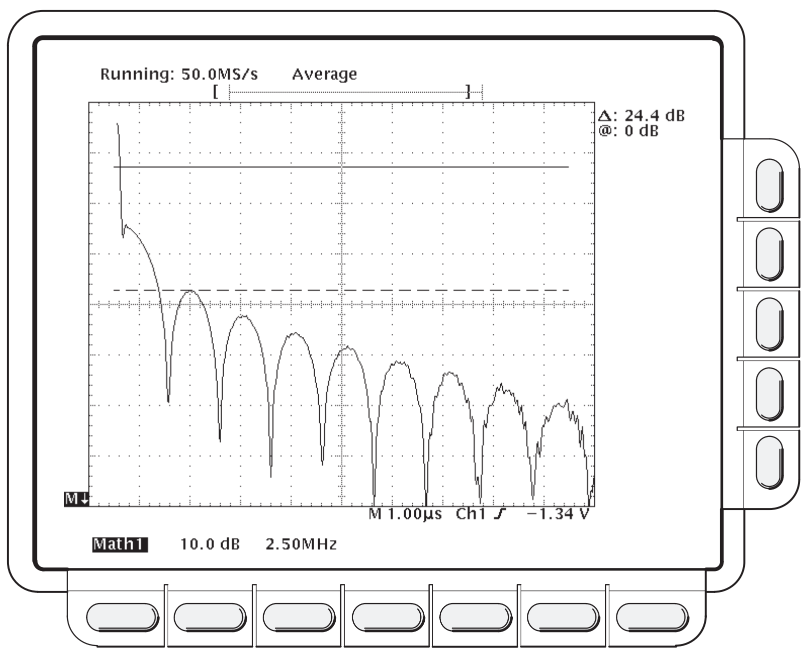


Figure 1-4: Cursor Measurement of an FFT Waveform

10. Press **Function** (main) → **Paired** (side).
11. Use the technique just outlined to place the vertical bar of each paired cursor to the points along the horizontal axis you are interested in.

12. Read the amplitude between the short horizontal bar of the two paired cursors from the top-most **Δ:** readout. Read the amplitude of the short horizontal bar of the selected (solid) cursor relative to either 1 V_{RMS} (0 dB), ground (0 volts), or zero phase level (0 degrees or 0 radians) from the **@:** readout. Read the frequency between the long horizontal bars of both paired cursors from the bottom **Δ:** readout.

Automated Measurements of an FFT

You can also use automated measurements to measure FFT math waveforms. Use the same procedure as is found under *Waveform Differentiation* on page 2-2.

Considerations for Using FFTs

There are several characteristics of FFTs that affect how they are displayed and should be interpreted. Read the following topics to learn how to optimize the oscilloscope setup for good display of your FFT waveforms.

The FFT Frequency Domain Record

The following topics discuss the relation of the source waveform to the record length, frequency resolution, and frequency range of the FFT frequency domain record. (The FFT frequency domain waveform is the FFT math waveform that you display.)

FFTs May Not Use All of the Waveform Record—The *FFT math waveform* is a display of the magnitude or phase data from the *FFT frequency domain record.* This frequency domain record is derived from the FFT time domain record, which is derived from the *waveform record.* All three records are described below.

Waveform Record—the complete waveform record acquired from an input channel and displayed from the same channel or a reference memory. The length of this *time domain* record is user-specified from the Horizontal menu. The waveform record is not a DSP Math waveform.

FFT Time Domain Record—that part of the waveform record that is input to the FFT. This time domain record waveform becomes the FFT math waveform after it is transformed. Its record length depends on the length of the waveform record defined above.

FFT Frequency Domain Record—the FFT math waveform after digital signal processing converts data from the FFT time domain record into a frequency domain record.

Figure 1-5 compares the waveform record to the FFT time domain record. Note the following relationships:

- For waveform records ≤10 K points in length, the FFT uses all of the waveform record as input.

- For waveform records >10 K points, the first 10 K points of the waveform record becomes the FFT time domain record.
- Each FFT time domain record starts at the beginning of the acquired waveform record.
- The zero phase reference point for a phase FFT math waveform is in the middle of the FFT time domain record regardless of the waveform record length.

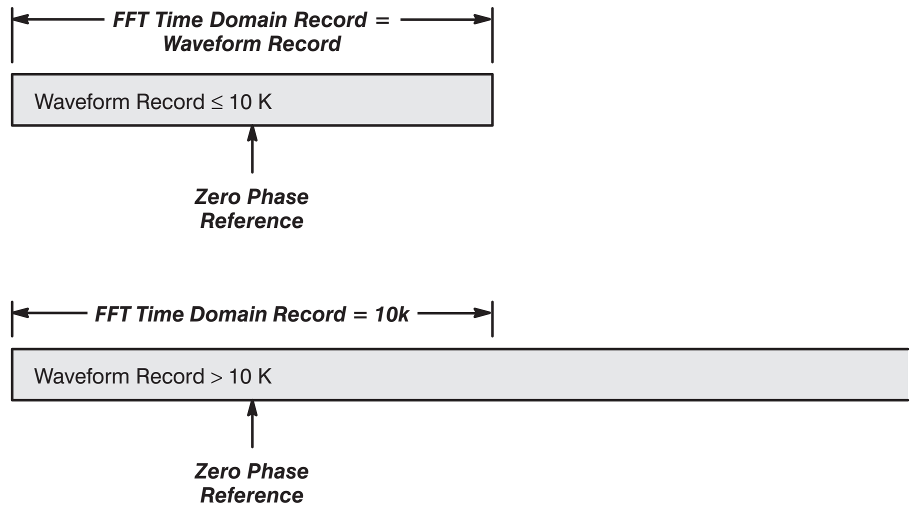


Figure 1-5: Waveform Record vs. FFT Time Domain Record

FFTs Transform Time Records to Frequency Records—The FFT time domain record just described is input for the FFT. Figure 1-6 shows the transformation of that time domain data record into an FFT frequency domain record. The resulting frequency domain record is one half the length of the FFT input because the FFT computes both positive and negative frequencies. Since the negative values mirror the positive values, only the positive values are displayed.

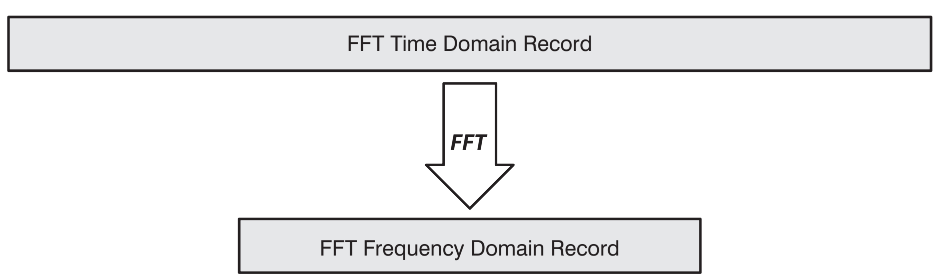


Figure 1-6: FFT Time Domain Record vs. FFT Frequency Domain Record

FFT Frequency Range and Resolution—When you turn on an FFT waveform, the oscilloscope displays either the magnitude or phase angle of the FFT frequency domain record. The resolution between the discrete frequencies displayed in this waveform is determined by the following equation:

$$\Delta F = \frac{Sample\ \ Rate}{FFT\ \ Length}$$

Where: *ΔF is the frequency resolution.*

Sample Rate is the sample rate of the source waveform.

FFT Length is the length of the FFT Time Domain waveform record.

The sample rate also determines the range these frequencies span; they span from 0 to ½ the sample rate of the waveform record. (The value of ½ the sample rate is often referred to as the Nyquist frequency or point.) For example, a sample rate of 20 Megasamples/second would yield an FFT with a range of 0 to 10 MHz. The sample rates available for acquiring data records vary over a range the limits of which depend on your oscilloscope model. TDS oscilloscopes (except for TDS 800 models) display the sample rate in the acquisition readout at the top of the oscilloscope screen.

Offset, Position, and Scale

The following topics contain information to help you display your FFT properly.

Adjust for a Non-Clipped Display—To properly display your FFT waveform, scale the source waveform so it is not clipped.

- You should scale and position the source waveform so it is contained on screen. (Off screen waveforms may be "clipped," which will result in errors in the FFT waveform).

  Alternately, to get maximum vertical resolution, you can display source waveforms with amplitudes up to two divisions greater than that of the screen. If you do, turn on **Pk-Pk** in the measurement menu and monitor the source waveform for clipping.

- Use vertical position and vertical offset to position your *source* waveform. As long as the source waveform is not clipped, its vertical position and vertical offset will not affect your FFT waveform except at DC. (DC correction is discussed below.)

Adjust Offset and Position to Zero for DC Correction—Normally, the output of a standard FFT computation yields a DC value that is twice as large as it should be with respect to the other frequencies. Also, the selection of window type introduces errors in the DC value of an FFT.

The displayed output of the FFT on TDS oscilloscopes is corrected for these errors to show the true value for the DC component of the input signal. The **Position** and **Offset** must be set to zero for the source waveform in the Vertical menu. When measuring the amplitude at DC, remember that 1 VDC equals 1 V_{RMS} and the display is in dB.

Record Length

Most often, you will want to use a short record length because more of the FFT waveform can be seen on screen and long record lengths can slow oscilloscope response. However, long record lengths lower the noise relative to the signal and increase the frequency resolution for the FFT. More important, they might be needed to capture the waveform feature you want to include in the FFT.

To speed up oscilloscope response when using long record lengths, you can save your source waveform in a reference memory and perform an FFT on the saved waveform. That way the DSP will compute the FFT based on saved, static data and will only update if you save a new waveform.

Acquisition Mode

Selecting the right acquisition mode can produce less noisy FFTs.

Set up in Sample or Normal Mode—Use sample mode (all models *except* TDS 800 Oscilloscopes) until you have set up and turned on your FFT. Sample mode can acquire repetitive and nonrepetitive waveforms and does not affect the frequency response of the source waveform.

Use normal mode if your oscilloscope is a TDS 800 Oscilloscope (sample mode is not available). Like sample mode for other models, normal mode does not affect the frequency response; however, unlike sample mode, it can only acquire repetitive waveforms.

Hi Res and Average Reduce Noise—After the FFT is set up and displayed, it might be useful to turn on Hi Res mode, on TDS models so equipped, to reduce the effect of noise in the signal. Hi Res operates on both repetitive and nonrepetitive waveforms; however, it does affect the frequency response of the source waveform.

If the pulse is repetitive, then Average mode may be used to reduce noise in the signal at a cost of slower display response. Average operates on repetitive waveforms only, and averaging does effect the frequency response of the source waveform.

Peak Detect (on TDS models so equipped) and Envelope mode can add significant distortion to the FFT results and are not recommended for use with FFTs.

Zoom and Interpolation

Once you have your waveform displayed optimally, you may magnify (or reduce) it vertically and horizontally to inspect any feature you desire. Just be sure the FFT waveform is the selected waveform. (Press **MORE**, then select the FFT waveform in the More main menu. Then use the Vertical and Horizontal **SCALE** knobs to adjust the math waveform size.)

If you wish to see the zoom factor (2X, 5X, etc.) you need to turn Zoom on: press **ZOOM** → **On** (side). The vertical and horizontal zoom factors appear on screen.

Whether Zoom is on or off, you can press **Reset Zoom Factors** (side) to return the zoomed FFT waveform to no magnification.

Zoom always uses either sin(x)/x or linear interpolation when expanding displayed waveforms. To select the interpolation method: press **DISPLAY** → **Filter** (main) → **Sin(x)/x** or **Linear** (side).

If the source waveform record length is 500 points, the FFT will use 2X Zoom to increase the 250 point FFT frequency domain record to 500 points. Therefore, FFT math waveforms of 500 point waveforms are always zoomed 2X or more with interpolation. Waveforms with other record lengths can be zoomed or not and can have minimum Zooms of 1X or less.

Sin(x)/x interpolation may distort the magnitude and phase displays of the FFT depending on which window was used. You can easily check the effects of the interpolation by switching between sin(x)/x and linear interpolation and observing the difference in measurement results on the display. If significant differences occur, use linear interpolation.

Undersampling (Aliasing)

Aliasing occurs when the oscilloscope acquires a source waveform with frequency components outside of the frequency range for the current sample rate. In the FFT waveform, the actual higher frequency components are undersampled, and therefore, they appear as lower frequency aliases that “fold back” around the Nyquist point (see Figure 1-7).

The greatest frequency that can be input into any sampler without aliasing is ½ the sample frequency. Since source waveforms often have a fundamental frequency that does not alias but have harmonic frequencies that do, you should have methods for recognizing and dealing with aliases:

- Be aware that a source waveform with fast edge transition times creates many high frequency harmonics. These harmonics typically decrease in amplitude as their frequency increases.
- Sample the source signal at rates that are at least 2X that of the highest frequency component having significant amplitude.
- Filter the input to bandwidth limit it to frequencies below that of the Nyquist frequency.

- Recognize and ignore the aliased frequencies.

If you think you have aliased frequencies in your FFT, select the source channel and adjust the horizontal scale to increase the sample rate. Since you increase the Nyquist frequency as you increase the sample rate, the alias signals should “unfold” and appear at their proper frequency.

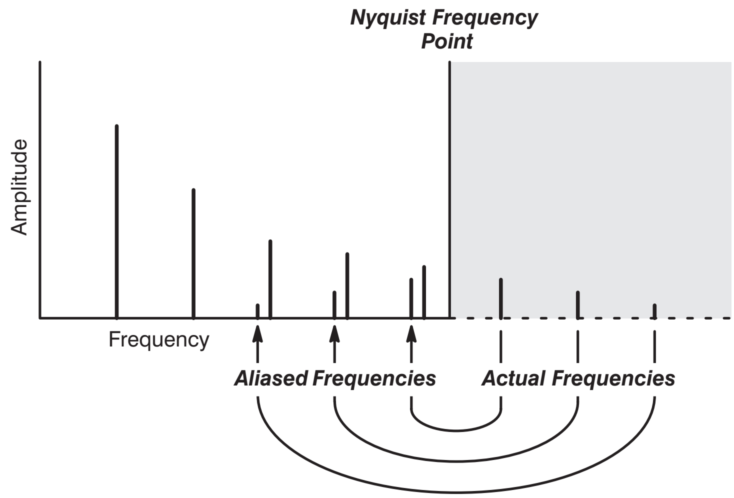


Figure 1-7: How Aliased Frequencies Appear in an FFT

Considerations for Phase Displays

When you setup an FFT math waveform to display the phase angle of the frequencies contained in a waveform, you should take into account the reference point the phase is measured against. You may also need to use phase suppression to reduce noise in your FFTs.

Establish a Zero Phase Reference Point—The phase of each frequency is measured with respect to the zero phase reference point. The zero reference point is the point at the center of the FFT math waveform but corresponds to various points on the *source* (time domain) record. (See Figure 1-5 on page 1-8.)

To measure the phase relative to most source waveforms, you need only to center the positive peak around the zero phase point. (For instance, center the positive half cycle for a sine or square wave around the zero phase point.) Use the following method:

- First be sure the FFT math waveform is selected in the More menu, then set horizontal position to 50% in the Horizontal menu. This positions the zero phase reference point to the horizontal center of the screen.

- In the Horizontal menu, vary the trigger position (time base position for TDS 800 model oscilloscopes) to center the positive peak of the source waveform at the horizontal center of screen. Alternately, you can adjust the trigger level (knob) to bring the positive peak to center screen if the phase reference waveform has slow enough edges.

When impulse testing and measuring phase, align the impulse input into the system to the zero reference point of the FFT time domain waveform:

- Set the trigger position (time base position for TDS 800 model oscilloscopes) to 50% and horizontal position to 50% for all record lengths less than 15 K. (Your model oscilloscope may not have record lengths of 15 K or longer—consult your User manual.)
- For records with a 15 K length, set the trigger position to 33% for all models *except TDS 800 oscilloscopes*. Use the horizontal position knob to move the trigger T on screen to the center horizontal graticule line.

  With TDS 800 oscilloscopes only, first set the record length to 5000 points in 1000 divisions. Then turn the horizontal position knob full counter clockwise, set the record length to 15000 points in 300 divisions, and adjust the time base position to move the impulse to the center horizontal graticule line.
- For records with 30 K, 50 K, or 60 K lengths (not all lengths are available for all TDS models—consult your User manual), set the trigger position to 16.6%,10%, or 8.3%, respectively. Use the horizontal position knob to move the trigger T on screen and to the center horizontal graticule line.
- Trigger on the input impulse.

Adjust Phase Suppression—Your source waveform record may have a noise component with phase angles that randomly vary from –pi to pi. This noise could make the phase display unusable. In such a case, use phase suppression to control the noise.

You specify the phase suppression level in dB with respect to 1 V_{RMS}. If the magnitude of the frequency is greater than this threshold, then its phase angle will be displayed. However, if it is less than this threshold, then the phase angle will be set to zero and be displayed as zero degrees or radians. (The waveform reference indicator at the left side of the graticule indicates the level where phase is zero for phase FFTs.)

It is easier to determine the level of phase suppression you need if you first create a *frequency* FFT math waveform of the source and then create a *phase* FFT waveform of the same source. Do the following steps to use a cursor measurement to determine the suppression level:

1. Do steps 1 through 7 of *Display an FFT* that begins on page 1-2. Select **dBV RMS** (side) for the **Set FFT Vert Scale to** (side).

2. Press **CURSOR** → **Mode** (main) → **Independent** (side) → **Function** (main) → **H Bars** (side). Use the general purpose knob to align the selected cursor to a level that places the tops of the magnitudes of frequencies of interest above the cursor but places other magnitudes completely below the cursor.
3. Read the level in dB from the **@:** readout. Note the level for use in step 5.
4. Press **MORE** (main) → **Change Waveform Definition menu** (side). **Press Set FFT Vert Scale to** (side) repeatedly to choose either **Phase (rad)** or **Phase (deg)**.
5. Press **Suppress Phase at Amplitudes** (side). Use the general purpose knob (or keypad if your oscilloscope is so equipped) to set phase suppression to the value obtained using the H Bar cursor. Do not change the window selection or you will invalidate the results obtained using the cursor.

FFT Windows

To learn how to optimize your display of FFT data, read about how the FFT windows data before computing the FFT math waveform. Understanding FFT windowing can help you get more useful displays.

Windowing Process—The oscilloscope multiplies the FFT time domain record by one of four FFT windows it provides before it inputs the record to the FFT function. Figure 1-8 shows how the time domain record is processed.

The FFT windowing acts like a bandpass filter between the FFT time domain record and the FFT frequency domain record. The shape of the window controls the ability of the FFT to resolve (separate) the frequencies and to accurately measure the amplitude of those frequencies.

Selecting a Window—You can select your window to provide better frequency resolution at the expense of better amplitude measurement accuracy in your FFT, better amplitude accuracy over frequency resolution, or to provide a compromise between both. You can choose from these four windows: Rectangular, Hamming, Hanning, and Blackman-Harris.

In step 8 (page 1-4) in *Displaying an FFT,* the four windows are listed in order according to their ability to resolve frequencies versus their ability to accurately measure the amplitude of those frequencies. The list indicates that the ability of a given window to resolve a frequency is inversely proportional to its ability to accurately measure the amplitude of that frequency. In general, then, choose a window that can just resolve between the frequencies you want to measure. That way, you will have the best amplitude accuracy and leakage elimination while still separating the frequencies.

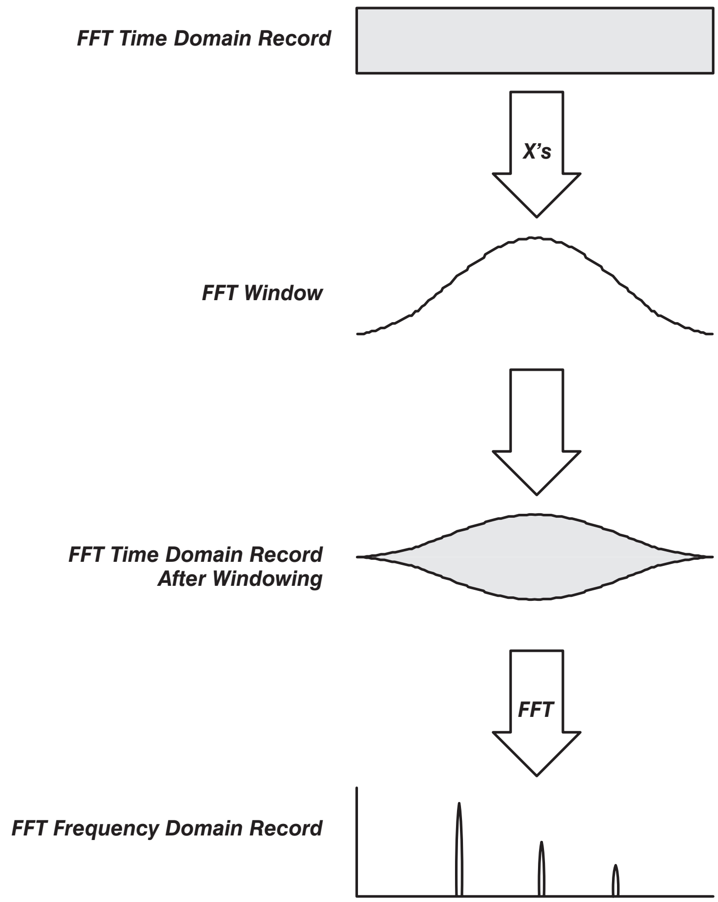


Figure 1-8: Windowing the FFT Time Domain Record

You can often determine the best window empirically by first using the window with the most frequency resolution (rectangular), then proceeding toward that window with the least (Blackman-Harris) until the frequencies merge. Use the window just before the window that lets the frequencies merge for best compromise between resolution and amplitude accuracy.

NOTE

If the Hanning window merges the frequencies, try the Hamming window before settling on the rectangular window. Depending on the distance of the frequencies you are trying to measure from the fundamental, the Hamming window sometimes resolves frequencies better than the Hanning.

Window Characteristics—When evaluating a window for use, you may want to examine how it modifies the FFT time domain data. Figure 1-9 shows each window, its bandpass characteristic, bandwidth, and highest side lobe. Consider the following characteristics:

- The narrower the central lobe for a given window, the better it can resolve a frequency.
- The lower the lobes on the side of each central lobe are, the better the amplitude accuracy of the frequency measured in the FFT using that window.
- Narrow lobes increase frequency resolution because they are more selective. Lower side lobe amplitudes increases accuracy because they reduce leakage.

  Leakage results when the FFT time domain waveform delivered to the FFT function contains a non-integer number of waveform cycles. Since there are fractions of cycles in such records, there are discontinuities at the ends of the record. These discontinuities cause energy from each discrete frequency to "leak" over on to adjacent frequencies. The result is amplitude error when measuring those frequencies.

The rectangular window does not modify the waveform record points; it generally gives the best frequency resolution because it results in the most narrow lobe width in the FFT output record. If the time domain records you measured always had an integer number of cycles, you would only need this window.

Hamming, Hanning, and Blackman-Harris are all somewhat bell-shaped widows that taper the waveform record at the record ends. The Hanning and Blackman/Harris windows taper the data at the end of the record to zero; therefore, they are generally better choices to eliminate leakage.

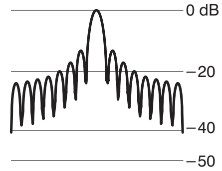
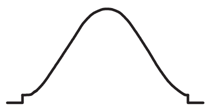
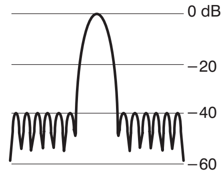
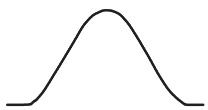
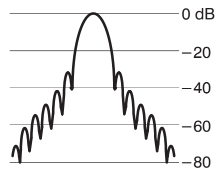
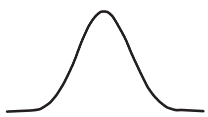
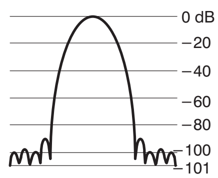
FFT Window Type	*Bandpass Filter*	*-3 dB Bandwidth*	*Highest Side Lobe*
 Rectangular Window		0.89	-13 dB
 Hamming Window		1.28	–43 dB
 Hanning Window		1.28	–32 dB
 Blackman-Harris Window		1.28	–94 dB

Figure 1-9: FFT Windows and Bandpass Characteristics

Care should be taken when using bell shaped widows to be sure that the most interesting parts of the signal in the time domain record are positioned in the center region of the window so that the tapering does not cause severe errors.

Waveform Differentiation

Waveform Differentiation

The Advanced DSP Math option provides *waveform differentiation* that allows you to display a derivative math waveform that indicates the instantaneous rate of change of the waveform acquired. Such waveforms are used in the measurement of slew rate of amplifiers and in educational applications. You can store and display a derivative math waveform in a reference memory, then use it as a source for another derivative waveform. The result is the second derivative of the waveform that was first differentiated.

Description

The math waveform, derived from the sampled waveform, is computed based on the following equation:

$$Y_n = (X_{(n+1)} - X_n)\frac{1}{T}$$

Where: *X is the source waveform*

Y is the derivative math waveform

T is the time between samples

Since the resultant math waveform is a derivative waveform, its vertical scale is in volts/second (its horizontal scale is in seconds). The source signal is differentiated over its entire record length; therefore, the math waveform record length equals that of the source waveform.

Operation

To obtain a derivative math waveform, do the following tasks:

- Acquire and display the waveform in your choice of input channels.
- Differentiate it using the math waveform menu.
- Use cursors or automated measurements to measure its parameters.

Use the following procedure to perform these tasks.

Displaying a Differentiated Waveform

1. Connect the waveform to the desired channel input and select that channel.
2. Adjust the vertical and horizontal scales and trigger the display (or press **AUTOSET**).

3. Press **MORE** → **Math1**, **Math2**, or **Math3** (main) → **Change Math Definition** (side) → **Single Wfm Math** (main). See Figure 1-2.

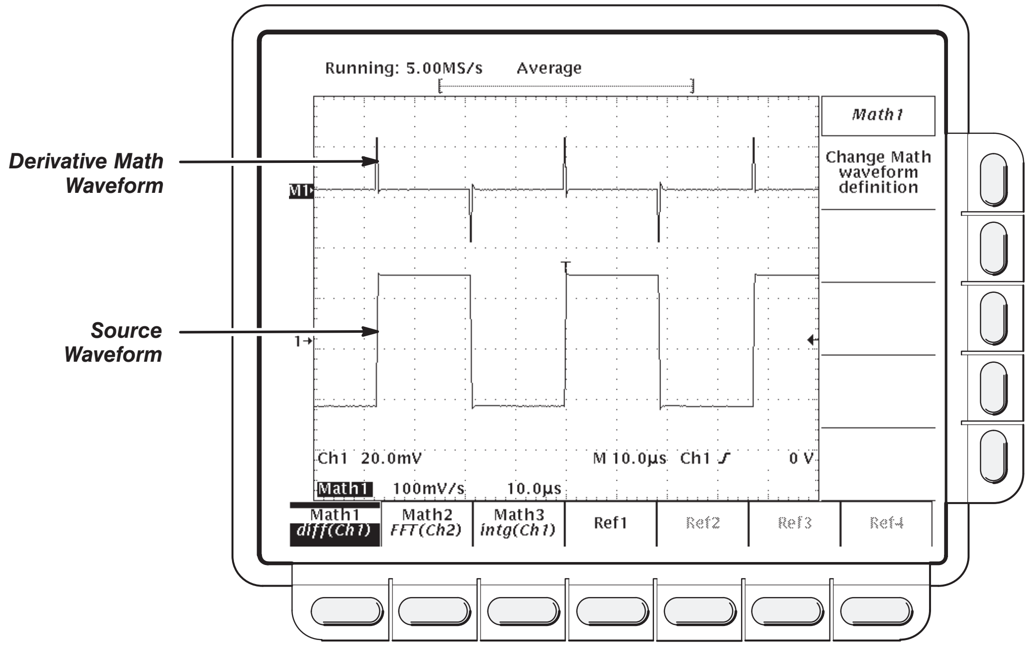


Figure 2-1: Derivative Math Waveform

4. Press **Set Single Source to** (side). Repeatedly press the same button (or use the general purpose knob) until the channel source selected in step 1 appears in the menu label.
5. Press **Set Function to** (side). Repeatedly press the same button (or use the general purpose knob) until **diff** appears in the menu label.
6. Press **OK Create Math Wfm** (side) to display the derivative of the waveform you input in step 1.

   You should now have your derivative math waveform on screen. Use the Vertical **SCALE** and **POSITION** knobs to size and position your waveform as you require.

Automated Measurements of a Derivative Waveform

Once you have displayed your derivative math waveform, you can use automated measurements to make various parameter measurements. Do the following steps to display automated measurements of the waveform:

1. Be sure **MORE** is selected in the channel selection buttons and that the differentiated math waveform is selected in the **More** main menu.
2. Press **MEASURE** → **Select Measrmnt** (main).

3. Select up to four measurements in the side menu. (Push **More** (side) to see more measurement choices.) See your User Manual for descriptions of each measurement.

   You can press **Remove Measrmnts** (main) to display a side menu for removing measurements from the display.

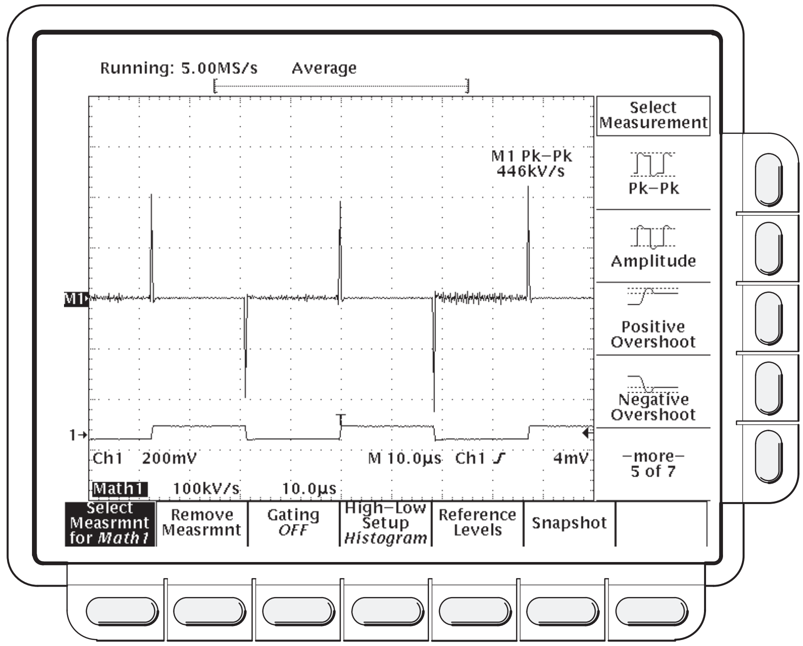


Figure 2-2: Peak-Peak Amplitude Measurement of a Derivative Waveform

You can also get a snapshot of all single waveform measurements at once:

4. Press **Snapshot** (main). Most of the measurements found in the Measure side menus are now in a single display.

Cursor Measurement of a Derivative Waveform

You can also use cursors to measure derivative waveforms. Use the same procedure as is found under *Waveform Integration* on page 3-2. When using that procedure, note that the amplitude measurements on a derivative waveform will be in volts/second rather than in volt-seconds as is indicated for the integral waveform measured in the procedure.

Usage Considerations

When creating differentiated math waveforms from live channel waveforms, consider the following topics.

Offset, Position, and Scale

Note the following tips for obtaining a good display:

- You should scale and position the source waveform so it is contained on screen. (Off screen waveforms may be "clipped," which will result in errors in the derivative waveform).
- You can use vertical position and vertical offset to position your *source* waveform. The vertical position and vertical offset will not affect your derivative waveform unless you position the source waveform off screen so it is clipped.
- When using the vertical scale knob to scale the source waveform, note that it also scales your derivative waveform.

Because of the method the oscilloscope uses to scale the source waveform before differentiating that waveform, the derivative math waveform may be too large vertically to fit on screen—even if the source waveforms is only a few divisions on screen. You can use Zoom to reduce the size of the waveform on screen (see *Zoom* that follows), but If your waveform is clipped before zooming, it will still be clipped after it is zoomed.

If your math waveform is a narrow differentiated pulse, it may not appear to be clipped when viewed on screen. You can detect if your derivative math waveform is clipped by expanding it horizontally using Zoom so you can see the clipped portion. Also, the automated measurement **Pk-Pk** will display a clipping error message if turned on (see *Automated Measurements of a Derivative Waveform* on page 2-2).

If your derivative waveform is clipped, try either of the following methods to eliminate clipping:

- Reduce the size of the source waveform on screen. (Select the source channel and use the vertical **SCALE** knob.)
- Expand the waveform horizontally on screen. (Select the source channel and increase the horizontal scale using the horizontal **SCALE** knob.) For instance, if you display the source waveform illustrated in Figure 2-1 on page 2-2 so its rising and falling edges are displayed over more horizontal divisions, the amplitude of the corresponding derivative pulse will decrease.

Whichever method you use, be sure Zoom is off and the zoom factors are reset (see *Zoom* below).

Zoom

Once you have your waveform optimally displayed, you can also magnify (or contract) it vertically and horizontally to inspect any feature. Just be sure the differentiated waveform is the selected waveform. (Press **MORE**, then select the differentiated waveform in the More main menu. Then use the Vertical and Horizontal **SCALE** knob to adjust the math waveform size.)

If you wish to see the zoom factor (2X, 5X, etc.), you need to turn zoom on: press **ZOOM → ON** (side). The vertical and horizontal zoom factors appear on screen.

Whether zoom is on or off, you can press **Reset Zoom Factors** (side) to return the zoomed derivative waveform to no magnification.

Waveform Integration

Waveform Integration

The Advanced DSP Math option provides *waveform integration* that allows you to display an integral math waveform that is an integrated version of the acquired waveform. Such waveforms find use in the following applications:

- Measuring of power and energy, such as in switching power supplies
- Characterizing mechanical transducers, as when integrating the output of an accelerometer to obtain velocity

Description

The integral math waveform, derived from the sampled waveform, is computed based on the following equation:

$$y(n) = scale \sum_{i=1}^{n} \frac{x(i) + x(i-1)}{2} T$$

Where: *x(i) is the source waveform*

y(n) is a point in the integral math waveform

scale is the output scale factor

T is the time between samples

Since the resultant math waveform is an integral waveform, its vertical scale is in volt-seconds (its horizontal scale is in seconds). The source signal is integrated over its entire record length; therefore, the math waveform record length equals that of the source waveform.

Operation

To obtain an integral math waveform, you first acquire it in your choice of input channels. Then you integrate it by creating an integral math waveform from the math waveform menu. Once the math waveform is displayed, you can use cursors or automated measurements to measure its parameters.

Displaying an Integral Waveform

Do the following steps to integrate and display your waveform.

1. Connect the waveform to the desired channel input and select that channel.
2. Adjust the vertical and horizontal scales and trigger the display (or press **AUTOSET**).

3. Press **MORE** → **Math1**, **Math2**, or **Math3** (main) → **Change Math waveform definition** (side) → **Single Wfm Math** (main).
4. Press **Set Single Source to** (side). Repeatedly press the same button (or use the general purpose knob) until the channel source selected in step 1 appears in the menu label.
5. Press **Set Function to** (side). Repeatedly press the same button (or use the general purpose knob) until **intg** appears in the menu label.
6. Press **OK Create Math Waveform** (side) to turn on the integral math waveform.

   You should now have your integral math waveform on screen. See Figure 3-1. Use the Vertical **SCALE** and **POSITION** knobs to size and position your waveform as you require.

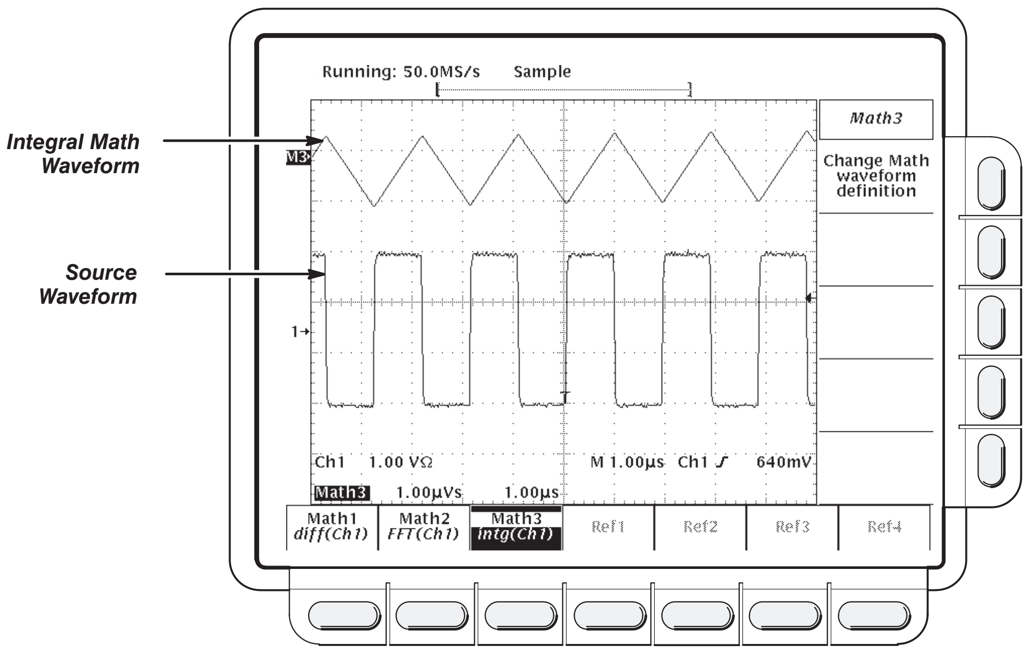


Figure 3-1: Integral Math Waveform

Cursor Measurements of an Integral Waveform

Once you have displayed your integrated math waveform, use cursors to measure its voltage over time.

1. Be sure **MORE** is selected (lit) in the channel selection buttons and that the integrated math waveform is selected in the **More** main menu.
2. Press **CURSOR** → **Mode** (main) → **Independent** (side) → **Function** (main) → **H Bars** (side).

3. Use the general purpose knob to align the selected cursor (solid) to the top (or to any amplitude level you choose).
4. Press **TOGGLE** to select the other cursor.
5. Use the general purpose knob to align the selected cursor (to the bottom (or to any amplitude level you choose).
6. Read the integrated voltage over time between the cursors in volt-seconds from the **Δ:** readout. Read the integrated voltage over time between the selected cursor and the reference indicator of the math waveform from the **@:** readout. See Figure 3-2.

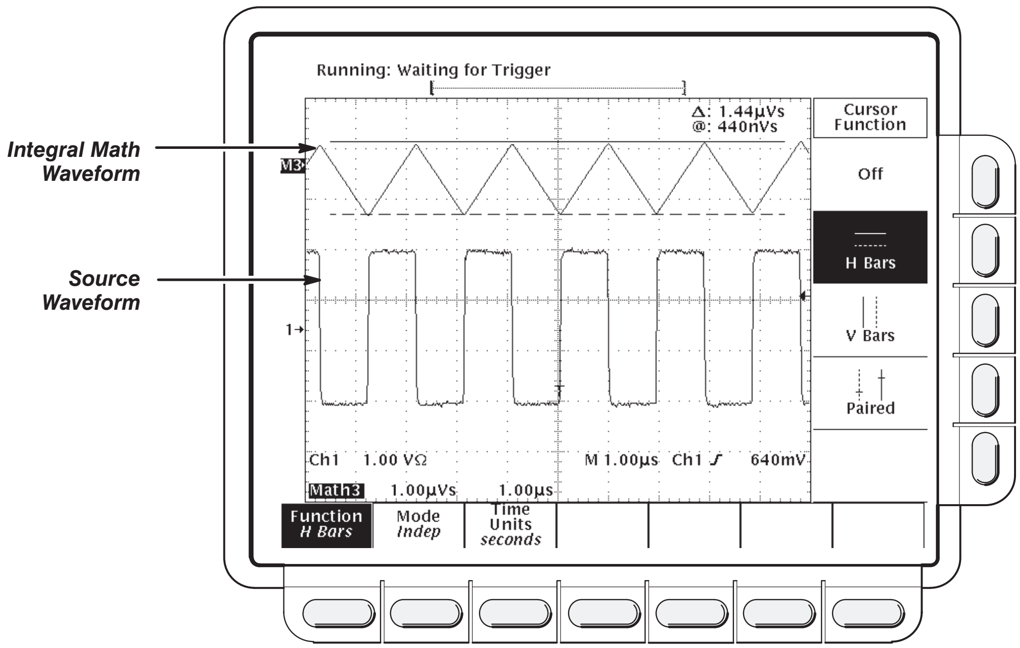


Figure 3-2: H Bars Cursors Measure an Integral Math Waveform

7. Press **Function** (main) ➞ **V Bars** (side).Use the general purpose knob to align one of the two vertical cursors to a point of interest along the horizontal axis of the waveform.
8. Press **TOGGLE** to select the alternate cursor.
9. Align the alternate cursor to another point of interest on the math waveform.
10. Read the time difference between the cursors from the **Δ:** readout. Read the time difference between the selected cursor and the trigger point for the *source* waveform from the **@:** readout.
11. Press **Function** (main) ➞ **Paired** (side).

12. Use the technique just outlined to place the long vertical bar of each paired cursor to the points along the horizontal axis you are interested in.

13. Read the following values from the cursor readouts:

    - Read the integrated voltage over time between the *short* horizontal bars of both paired cursors in volt-seconds from the **Δ:** readout.
    - Read the integrated voltage over time between the *short* horizontal bar of the selected cursor and the reference indicator of the math waveform from the **@:** readout.
    - Read the time difference between the *long* vertical bars of the paired cursors from the **Δ:** readout.

Automated Measurements of a Integral Waveform

You can also use automated measurements to measure integral math waveforms. Use the same procedure as is found under *Waveform Differentiation* on page 2-2. When using that procedure, note that your measurements on an integral waveform will be in volt-seconds rather than in volts/second as is indicated for the differential waveform measured in the procedure.

Usage Considerations

When creating integrated math waveforms from live channel waveforms, consider the following topics.

Offset, Position, and Scale

Note the following requirements for obtaining a good display:

- You should scale and position the source waveform so it is contained on screen. (Off screen waveforms may be “clipped,” which will result in errors in the integral waveform).
- You can use vertical position and vertical offset to position your source waveform. The vertical position and vertical offset will not affect your integral waveform unless you position the source waveform off screen so it is clipped.
- When using the vertical scale knob to scale the source waveform, note that it also scales your integral waveform.

DC Offset

The source waveforms that you connect to the oscilloscope often have a DC offset component. The oscilloscope integrates this offset along with the time varying portions of your waveform. Even a few divisions of offset in the source waveform may be enough to ensure that the integral waveform saturates (clips), especially with long record lengths.

You may be able to avoid saturating your integral waveform if you choose a shorter record length. (Press **HORIZONTAL MENU** → **Record Length** (main).) Reducing the sample rate (use the **HORIZONTAL SCALE** knob) with the source channel selected might also prevent clipping. You can also select AC coupling (on TDS models so equipped) in the vertical menu of the *source* waveform or otherwise DC filter it before applying it to the oscilloscope input.

Zoom

Once you have your waveform optimally displayed, you may magnify (or reduce) it vertically and horizontally to inspect any feature you desire. Just be sure the integrated waveform is the selected waveform. (Press **MORE**, then select the integrated waveform in the More main menu. Then use the Vertical and Horizontal **SCALE** knobs to adjust the math waveform size.)

If you wish to see the zoom factor (2X, 5X, etc.) you need to turn Zoom on: press **ZOOM** → **On** (side). The vertical and horizontal zoom factors appear on screen.

Whether Zoom is on or off, you can press **Reset Zoom Factors** (side) to return the zoomed integral waveform to no magnification.

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