CSA-1
2 MHz FFT Spectrum Analyzer

FFT Primer

2/02

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Belar History

Belar Electronics Laboratory, Inc. is the world's leading manufacturer of modulation monitors, which are used by radio and television stations to monitor important technical aspects of their broadcast signal. Belar also makes frequency monitors, specialized AM receiving antennas, and other related equipment for the broadcast industry.

Belar was founded in 1964 by the late Isobel ("BEL") and Arno ("AR") Meyer. Arno Meyer is still the President and owner. They started Belar in the basement of their house in Drexel Hill, PA, doing consulting and product design. The company moved to larger facilities in Upper Darby, PA in the late 60's, and then to its current location in Devon, PA in 1972.

The company was started to supply radio stations with the newly required FM stereo monitoring equipment, and later expanded to supply equipment needed by AM, shortwave, and TV broadcasters. Belar has always tried to offer products with the highest performance possible at a reasonable price. The Federal Communications Commission first used Belar monitors to verify their measurement of commercial FM broadcast stereo performance standards in the mid-1960's. Since then, Belar monitors have become industry standards, used as test instruments by many broadcast industry manufacturers (such as transmitter companies, etc.) to ensure that the products they make meet their specifications.

Belar products are also known for their durability—the vast majority are still in use today. Belar shipped its first model FMM-1 FM Monitor in 1966 to what is now WBEB 101.1, Philadelphia, where it is still in use as a back-up monitor.

In 2001, in recognition of Arno Meyer's and Belar's contribution to the industry, the National Association of Broadcasters awarded Arno Meyer with the coveted NAB Radio Engineering Achievement Award.

Belar markets its products domestically and internationally by direct sales and through distributors. Belar products are in use 24 hours a day, 7 days a week, in almost every country in the world.

FFT PRIMER

Introduction

The Belar CSA-1 uses digital signal processing (DSP) to perform its Fast Fourier Transform (FFT) functions. A process once prohibitively expensive, high-speed DSP has made the analysis power of FFT affordable in a standalone device. And because of the advent of high-speed, high-resolution A/D converters, it's now possible to directly analyze signals up to 2 MHz in real-time. Unlike software that requires a host PC, which may become distracted at times running other applications, the CSA-1's dedicated performance is predictable and reliable.

FFT is a mathematical routine that converts digital amplitude vs. time records into a more useful display of amplitude vs. frequency—the data is transformed from the "time domain" into the "frequency domain" without any loss of information. In the time domain, the signal is known as a waveform, but after conversion to the frequency domain, it is referred to as a spectrum. The same information is contained in each, and one can be converted into the other.

Although signals in the time domain have their uses, such as those viewed on an oscilloscope, it is often important to know how much energy exists at specific frequencies. That's where the frequency domain becomes important.

Basically speaking, an FFT spectrum analyzer divides the spectrum (the range of frequencies of interest) into bands, then measures and displays the amplitude level in each band. And because of its precision, it can do so accurately and with very high resolution in terms of both level and frequency.

Audio Analysis Comparison

Unlike the familiar fractional-octave audio spectrum analyzer—such as 1/3 octave—which spans a larger frequency range with increasing frequency because of a constant bandwidth percentage, FFT uses bands (or "bins" to be precise) that are equal in width. (An octave is a doubling of frequency—100 Hz to 200 Hz is one octave, just as 10 kHz to 20 kHz is one octave.)

Pink noise, which is white (random) noise that has been filtered by a special 3 dB per octave filter, will produce a "flat" display on a fractional-octave analyzer, but you will see a gentle (–3.0 dB per octave) high-frequency rolloff slope when using FFT. White noise, in contrast, will show a rising high frequency slope (+3.0 dB per octave) on a fractional-octave analyzer, but will appear to have a flat response when FFT analyzed. Stated differently, white noise has equal energy per Hertz whereas pink noise has equal energy per octave.

Because of this characteristic, high-frequency audio will not be as pronounced on an FFT when compared to a fractional-octave analyzer. It's important to understand,

though, that the FFT is not measuring audio incorrectly, just differently. And it's that difference, along with the increased amplitude and frequency resolution, that really distinguishes FFT.

A 1/3-octave analyzer usually divides the spectrum into 31 bands ranging from 20 Hz to 20 kHz, each having a bandwidth 23% of its center frequency. At the low end of the spectrum, where the first band is centered on 20 Hz, the bandwidth is 4.6 Hz, which is reasonable resolution. But at the opposite end—20 kHz—the bandwidth is 4.6 kHz. This very wide bandwidth does not allow for close scrutiny of response aberrations, or noise or distortion components. In addition, most dedicated analyzers are limited in their amplitude resolution (such as 1 dB) and range (perhaps 10 to 12 dB, without loosing resolution).

When an audio input is selected on the CSA-1—either analog or digital—you can select individual bins as large as 100 Hz, but as small as 1 Hz. In the 1 Hz mode, the CSA-1 divides the spectrum from 0 Hz (DC) to 24 kHz into 24,000 bins that are 1 Hz wide, then displays the level in each bin over a 120 dB range. This is precisely what makes FFT so powerful and desirable.

FFT in More Detail

Being a digitally based (sampled) device, it analyzes signals up to one-half of its sample rate. In the CSA-1, this translates into a bandwidth of 24 kHz for audio work, but is extended to 150 kHz, 1 MHz and 2 MHz, making it suitable for FM broadcast composite stereo through RF/IF work (radio frequency/intermediate frequency).

FFT bases its analysis on a snapshot of what it believes is exactly one cycle of a periodic (recurring) waveform. However, standard FFT applications are of varying, random signals. As in audio editing and AC switching applications, an ideal place to begin and end is at "zero crossing"—the point in the waveform where the voltage is zero. But because the snapshots are not synchronized to the signal being measured, the beginning of the FFT "record" (snapshot) could be at some positive voltage, and its end could be at a negative voltage. The difference between the picture's beginning and end, when analyzed, will appear as broadband noise that will mask low-level information, just as an audio edit not at zero crossing will produce a wideband "pop."

To solve this problem, "windows" are applied in the time domain (before the FFT) to more accurately depict what is being measured. The window tapers the block of data captured at the beginning and end of the record towards zero (it truncates the data towards zero at the edges). Exactly how the data is tapered is what makes the windows different. So you can think of a window as a filter—by removing the erroneous noise data, the relevant information is revealed.

Three FFT window traits to consider are amplitude response within the center bin, main or central lobe width, and "leakage." Although each window type will measure a signal's amplitude without error if its frequency is exactly centered within a bin, a

slight error can occur if the signal is at the bin's edge. Maximum error ranges from a modest –0.8 dB for Blackman-Harris (–92) to –3.9 dB for Rectangular.

Main lobe width affects selectivity, which is the ability to resolve closely spaced frequencies. An increase in main lobe width causes the vertical trace to widen. Rectangular is the narrowest; single tones will be displayed as vertical lines one "pixel" wide (a pixel is the smallest "picture element" a device can display). Blackman-Harris (–92) is the widest; a single tone will be 7 pixels in width.

There will always be "leakage"—some of the signal will leak into nearby bins at reduced level. This causes the noise floor to increase, which could obscure low-level signals, especially those that are adjacent. Rectangular, in general, has the highest leakage resulting in the highest noise floor, whereas Blackman-Harris has the least, producing the lowest noise floor. The maximum useable level range, or dynamic range, is dependant upon the noise floor.

Consequently, there is no one perfect window, just as there is no perfect filter. Each, though, has its application. In the CSA-1, there are 6 windows to choose from, based on 5 window types: Rectangular, Triangular, Hanning, Hamming, and Blackman-Harris. The simplest are those based on geometric shapes: Rectangular and Triangular. Those named after their creators—Hanning, Hamming, and Blackman-Harris—are more complex.

As you can see, FFT is an intricate process that involves a number of compromises. However, these compromises are not unique to the CSA-1, but are common to the FFT process.

Window Types

The response of each window type over a ±0.5 and ±10 bin range is illustrated in the graphs that follow. In all cases, bin centers are multiples of the Horizontal Resolution (HR).

Rectangular is perhaps the most basic window type and is very good for inspecting single frequencies that fall on bin centers, especially those that are closely spaced at bin multiples, because of its sharpness (deep nulls occur at ±1 bin).

Looking first at the magnified (dashed line) plot, it is 1 bin wide, from –0.5 to +0.5, is centered on 0, and uses the 0 to –6 dB response scale. Since the graph is plotted in "normalized frequency bins," the 0 bin point can represent any frequency from 0 Hz (DC) to 24 kHz for audio, and up to 2 MHz for extended range work. So if the horizontal resolution were set to 5 Hz, then the frequency range of the bin would be ±2.5 Hz (5 Hz x ±0.5).

As discussed above, if a tone falls exactly within the center of a bin, its level will be displayed without error. If the signal is offset by 0.1 bin high or low, it will be attenuated by –0.14 dB (assuming a 1 kHz tone and a horizontal resolution of 5 Hz, the offset would equal 0.5 Hz, as in 1.0005 kHz, or 1000.5 Hz). At 0.2, it increases

to –0.58 dB, and at the extreme edges of ±0.5 bin (±2.5 Hz in this example), it reaches its maximum of –3.9 dB.

The expanded ±10 bin graph (solid line) demonstrates response of the window over a wider range. It is also centered on 0, but uses the 120 dB scale. Remember that the center, or bin 0, extends to ±0.5. That means that the next bin, which is 1, runs from 0.5 to 1.5. Bin 2 is from 1.5 to 2.5, etc.

For a given signal, the indicated level in bin ±1 can be just –13 dB below bin 0 because of leakage. Move further away from bin 0 and leakage decreases to –29 dB at ±9.5 bins. Although it continues to taper, leakage causes a rather high noise floor across the spectrum. An exception is a frequency that's exactly centered. Because deep nulls fall on bin centers (1, 2, 3, etc.), a 1.000 kHz tone (HR: 5 Hz) will be seen as a 1 pixel wide display with a low noise floor (below –95 dB at 500 Hz & 1.50 kHz). Shift the frequency to 1.0025 kHz, however, and the noise floor will rise by about 45 dB to –50 dB at 500 Hz and 1.50 kHz.

Triangular is not as sharp as Rectangular—a single tone will display as 3 pixels in width. Still, it can differentiate among closely spaced frequencies near bin centers. Maximum bin-edge error is –1.8 dB. It, too, has a rather high noise floor, but is lower than Rectangular. Leakage into adjacent bins is –27 dB, which increases to –46 dB at ±9 bins. Triangular is the simplest way to taper the FFT's record edges to zero without widening the main lobe and reducing resolution.

Hanning is reasonably good at separating frequencies that are closely spaced because of its sharpness, but is not as narrow as Rectangular—like Triangular, single tones are 3 pixels in width. Bin-edge error is a reasonable –1.4 dB. Because of a lower noise floor, it is a good choice for low-level signals. Adjacent bin leakage is attenuated by –31 dB, rapidly increasing to –68 dB at ±9.5 bins. Like Rectangular, nulls fall on bin centers.

Hamming is similar to Hanning in that a single tone will be 3 pixels wide, yielding good selectivity. The error at bin-edge is –1.7 dB, and its noise floor is quite low. However, unlike Hanning, Hamming has low leakage—adjacent bins are –44 dB, remaining relatively constant to –47 dB at ±9.5 bins. This window is a good choice for viewing real-time spectrums of dynamic signals such as program material. It offers a very good compromise between main lobe width and leakage. Nulls also fall on bin centers.

Blackman-Harris (-67) is less sharp; a single tone will be displayed as 5 pixels wide (first nulls occur at ±3 bins). Signals at the edge of a bin are reduced by –1.1 dB. Its key advantages are low leakage (–73 dB in adjacent bins to –75 dB at ±9.5 bins), which yields a low noise floor. Some nulls occur on bin centers.

Blackman-Harris –92 has the widest main lobe of all—a single tone will be displayed as 7 pixels in width, possibly masking nearby signals (first nulls are at ±4 bins). However, because of its increased width, error at the edge of a bin is only –

0.8 dB, the lowest. Its primary advantage is a very low noise floor due to minimum leakage. Second adjacent bins are –92 dB, increasing to –97 dB at ±9.5 bins. This is the window of choice for making wide dynamic range measurements of steady state signals, such as harmonic analysis of distortion products. It will produce the lowest FFT noise floor and allow resolution of signals to a level of –100 dB. Some nulls also fall on bin centers.

Restated, Rectangular is the narrowest but has the highest noise floor and bin-edge error. Triangular is not quite as sharp but offers a slightly lower noise floor and reduced bin-edge error. Hanning is as sharp as Triangular but has lower leakage. Hamming is likewise as sharp, but even a lower noise floor because of reduced leakage. The Blackman-Harris windows are the widest but the produce the lowest noise floors. Blackman-Harris –92 offers the lowest bin-edge error of –0.8 dB.

The table below summarizes the key differences.

Window Type	Single Tone Width	Bin Edge Error	Max. Leakage
Rectangular –13	1 pixel	-3.92205 dB	-13.26262 dB
Triangular -27	3 pixels	-1.78678 dB	-26.52505 dB
Hanning –31	3 pixels	-1.39511dB	-31.47221 dB
Hamming –43	3 pixels	-1.72903 dB	-42.67454 dB
Blackman-Harris –67	5 pixels	-1.10364 dB	-67.54887 dB
Blackman-Harris –92	7 pixels	-0.80926 dB	-92.01593 dB

This information serves as a guide only. Since the CSA-1 allows for quick window type changes, choose the one that best displays what you are trying to analyze.

PROCEDURES AND APPLICATIONS

Audio Analysis Procedure

OVERVIEW — The Belar CSA-1 is a very powerful tool for analyzing audio signals and accepts balanced and unbalanced analog signals, and AES/EBU and S/PDIF digital audio signals. You can view one channel or two channels simultaneously, and select from left, right, L+R, or L-R. This allows for a variety of left and right signal parameter measurements including level, channel and spectral balance, separation, and crosstalk.

DETAILS — For audio work, you can select from Single or Dual modes. Single provides the greatest horizontal resolution because the entire width of the display is used. However, Dual allows two channels to be displayed simultaneously, side-by-side.

Connect either a balanced (XLR) or unbalanced (BNC) analog signal (because balanced and unbalanced inputs are summed, connect only one). The balanced input is bridging (12 kΩ). The unbalanced input is high impedance (100 kΩ), making it ideal for all types of unbalanced equipment even consumer grade (consumer equipment often specifies a minimum load impedance of 47 kΩ). The CSA-1 accepts full-scale input levels of –14 to +16 dBu (rear panel adjustable), covering "pro" (typically +4 dBu) and consumer levels (often –10 dBV = –7.8 dBu). For consumer work, BNC to RCA female adapters are readily available.

You can also connect either an AES/EBU (XLR) or a S/PDIF (coaxial, RCA jack) digital audio signal (connect either the AES/EBU or the S/PDIF). For S/PDIF optical (TOSLINK) applications, use an external optical to coaxial adapter. Sample rates from 10 to 56 kHz are supported (an internal sample rate converter automatically adapts the incoming signal to the 48 kHz sample rate used by the CSA-1). This wide range covers common applications: Broadcast (32 to 48 kHz), Consumer (44.1 kHz), and "pro" (48 kHz).

All bit depths may be used with the CSA-1. For 20 bits and above, the full 120 dB dynamic range will be available. Below 20 bits, it will be limited by the input signal. For example, the dynamic range of 16 bits is about 96 dB.

Both analog and digital signals can be connected simultaneously allowing you to select and view, say, a left channel analog channel, and a right channel digital channel.

The basic setup procedure is to:

1. Select input, Single or Dual

2. Select input source, analog (A/D) or digital (AES/EBU, which includes S/PDIF)

3. Adjust Vertical scale (dB) and Vertical Resolution (VR)

4. Adjust Horizontal Resolution (HR)

5. Select Display Mode

6. Select Window Type

Press MENU SEL and use the MENU ∧ ∨ keys to highlight INPUT SELECT, and then press SEL. Choose from Single or Dual. If Single, you will select one input; if Dual, you will make a selection for Channel #1 and one for Channel #2 (then choose DONE when finished). You may select from:

- 24 kHz L A/D — Selects the left channel analog input (XLR and BNC).

- 24 kHz R A/D — Selects the right channel analog input (XLR and BNC).

- 24 kHz L+R A/D — This option provides an L+R signal (left and right channels summed to mono) from the left and right analog inputs (XLR and BNC).

- 24 kHz L-R A/D — This choice produces an L-R signal (right channel subtracted from the left channel), which shows the "difference" between the two. If the two channels are very different (there is a great deal of stereo information), the L-R level will be very high. If the left and right channels are identical, which would occur if the signal were mono, the L-R level will be very low (under ideal conditions it will be below –120 dB).

  To achieve the deepest null (maximum subtraction or cancellation), the input levels (L ADJ and R ADJ on the rear panel) must be precisely matched. A good way to balance the inputs is to feed a mono signal, such as a tone or noise source (like white or pink), to the L and R inputs of the CSA-1. Display 24 kHz L-R A/D, and adjust the vertical resolution so that –120 dB can be observed. Then adjust either L ADJ or R ADJ (only one) for the deepest null (lowest level). The null will be very sharp, so adjust the control slowly.

- 24 kHz L AES/EBU — Selects the left channel of the AES/EBU (XLR) or S/PDIF (RCA) digital audio signal.

- 24 kHz R AES/EBU — Selects the right channel of the AES/EBU (XLR) or S/PDIF (RCA) digital audio signal.

- 24 kHz L+R AES/EBU — Like L+R A/D above, this choice provides an L+R (left and right channels summed to mono) signal from the AES/EBU (XLR) or S/PDIF (RCA) digital audio input.

- 24 kHz L-R AES/EBU — Like L-R A/D above, this option produces an L-R (right channel subtracted from the left channel) signal, which shows the "difference" between the two channels. If the two channels are very different (there is a great deal of stereo information), the L-R level will be very high. If the left and right channels are identical, which would occur if the signal were mono, the L-R level will be very low (under ideal conditions it will be below –120 dB). (Since it is a digital audio source, L ADJ and R ADJ do not affect null depth.)

In the Dual (two channel) mode, typical applications would be to display the left channel as Channel #1 and the right as Channel #2; or L+R as Channel #1, and L-R as Channel #2. However, the flexibility of the CSA-1 allows you to display any two of 8 choices.

Next, adjust the range and resolution. And, if desired, set the cursors…

- dB — Press VERT SEL until dB is highlighted, then use the VERT ∧ ∨ keys to select from +100 to –120 dB. For real-time audio work, it's common to place 0 at the top of the display. But since the reference level can be changed, any point within this range can be used as a reference.

- VR — Press VERT SEL until VR (vertical resolution) is highlighted, and use the VERT ∧ ∨ keys to select a resolution of 0.1 dB to 2.5 dB. Since the vertical resolution is 48 points, 2.5 dB allows display of the full 120 dB span, whereas 0.1 dB will display a range of 4.8 dB. There are a total of 7 choices. When viewing audio in real-time, a VR of 1.5 dB (72 dB) or 2.0 (96 dB) work well.

- C1 — Press VERT SEL until C1 (Cursor 1) is highlighted, and use the VERT ∧ ∨ keys to position the vertical cursor (the horizontal dotted line) to any one of 48 vertical points. This can be used as a reference to make it easy to see when levels exceed a certain threshold.

- HR — Press HORZ SEL until HR (horizontal resolution) is highlighted, and use the HORZ ∧ ∨ keys to select from a resolution of 1 Hz to 100 Hz in Single mode, or 1 Hz to 250 Hz in Dual mode.

- C2 — Press HORZ SEL until C2 (Cursor 2) is highlighted, and use the HORZ ∧ ∨ keys to set the position of the horizontal cursor (the vertical dotted line) to any one of 80 horizontal points in Dual, or any one of 200 horizontal points in Single. The exact level at the cursor location will appear under the frequency (top right-hand corner of the display). In Dual mode, the top level is Channel #1, the middle is Channel #2, and the bottom reading is the difference between Channels #1 and #2. Measurements are in dB relative to a reference you choose.

Set the Display Mode. The four options are described below:

- REALTIME — This mode shows the signal in "real-time"; the spectrum display is updated as fast as the FFT is calculated. The update rate varies from 100 milliseconds to 1 second depending on the selected frequency resolution. If the input signal is a continuous tone, the display will show its level and frequency without variation. If the input is audio program material, the display will vary constantly. Use this mode to observe the signal as it changes.

- PEAK HOLD — This choice continuously "monitors" the signal by continuing to calculate FFTs at the real-time rate for a specified period, then displays the highest level at each frequency during that period. It resets (starts over) at selectable time intervals. Choose this mode to "capture" and display snapshots of peaks of a varying signal, especially if the display is updating too quickly in Realtime mode. The update (reset)

rate can be set in 0.1 second increments up to 10 seconds. Unlike Infinite Hold, lower levels can replace higher levels.

- AVERAGE — This option, as its name suggests, "averages" a specified number of FFT samples (from 2 to 64). It is useful for displaying average levels rather than peak, just as a VU meter tends to display a rough "average" value. It increases measurement signal-to-noise ratio and can give an idea of perceived loudness of a signal.

- INFINITE HOLD — Similar to Peak Hold, Infinite Hold displays the highest (peak) level calculated in each bin. It accumulates until it is disabled or reset (it does not reset automatically like Peak Hold). A higher level will replace a lower level, but not vice versa. Then use the horizontal cursor (C2) to determine the level in bins of interest.

Finally, select the Window type. As discussed in detail in Window Types above, choose the window that presents the best display of your signal. Use the chart below as a general guide:

Window Type	Selectivity	Bin Edge Error	Noise Floor
Rectangular –13	High	High	High
Triangular -27	Medium	Medium	Medium
Hanning –31	Medium	Medium	Medium
Hamming –43	Medium	Medium	Medium
Blackman-Harris –67	Low	Low	Low
Blackman-Harris –92	Very Low	Very Low	Very Low

Again, because each signal is different, you may find it necessary to change the Window type to optimize the display, which is very quick on the CSA-1.

APPLICATIONS — The CSA-1 is a very flexible device that has applications beyond diagnostics. Some of the more common examples are listed below. Use an audio signal generator, a test CD and a CD player (which can be used for a source of digital test signals), or software that can generate test signals as a test source.

NOTE: Popular test CDs are: CBS Records Standard Test Disc, CD-1; Denon Audio Technical CD, C39-7147; and the NAB Broadcast and Audio System Test CD, and CD II. The NAB test CDs, in particular, feature a variety of useful broadcast-related test signals.

STUDIO APPLICATIONS

- Real-time level vs. frequency measurements of any two channels—left, right, L+R, or L-R, analog or digital. Channel balance and frequency

response differences can be quickly analyzed even using program material. The Average display mode is useful for general "loudness" indications.

- Measure even the most subtle input/output "difference" of any equalizer, compressor, limiter, or other processor, in real-time. Bridge the CSA-1's left channel across the input of the device under test, and connect the CSA-1's right channel to the device's output. Configure the CSA-1 for L-R operation. You can now view the dynamic level vs. frequency differences of the device.

- Test "lossy" audio compression algorithms such as MPEG Layer-3 (MP3). Use the infinite hold function to measure a song before and after processing. Even in the case of 128 Kbps, 44.1 kHz, stereo, constant bit rate encoding (11.0:1 compression)—the most common used on the Internet for "high quality" purposes—measure how high frequency information is discarded forever.

TEST & MEASUREMENT

- Check frequency response of a device or an entire system using one or more tones, a swept sine wave, or broadband noise. The Infinite Hold option will temporarily store the level at each frequency especially when using a swept sine wave or broadband noise. Use a calibrated or high-quality microphone and mic preamp for acoustic measurements. If you use broadband noise, be sure to use white noise instead of pink for a spectrally flat display, otherwise a –3.0 dB per octave slope will result.

- Distortion and harmonic analysis:

  - CCIF Twin-Tone intermodulation distortion (IMD) is a test that uses two tones spaced 1 kHz apart (typically 13 and 14 kHz for 15 kHz devices, and 19 and 20 kHz for 20 kHz units) to stress-test devices at their high frequency limit. Use the CSA-1 to measure the distortion products generated, especially those at the difference frequency of 1 kHz.

  - Inspect the harmonic distortion components of a single tone, or multiple, non-harmonically related tones.

  - Connect the CSA-1 to the monitor output of a distortion analyzer to view what is being measured (distortion products and noise).

- Examine the "noise floor" of a device or system, or study possible noise sources. An Average display mode will increase the apparent signal-to-noise ratio.

- Channel separation measurements between any two channels—L and R, analog and digital—are easy with the CSA-1. Configure it for Dual mode and use C2 (Cursor 2) to measure the level at each frequency, and the difference (separation) between the two channels. This can be done using a single tone, multiple tones, broadband noise, or program material. The use of program material allows for "dynamic," broadband separation tests under actual operating conditions.

- Crosstalk measurements between L+R and L-R channels are accomplished using the same procedure described in channel separation except that L+R and L-R are selected. And like channel separation, measurements can be made using one or more tones, broadband noise, or program audio.

High-Frequency Analysis Procedures

OVERVIEW — The Belar CSA-1 is not limited to just audio—or low frequency—measurements. High-speed, high-resolution A/D converters allow it to directly analyze signals up to 2 MHz in real-time. For broadcast, this includes AM transmitters, analog "composite" FM stereo signals, and—using a special down converter—FM transmitters too.

DETAILS — For high-frequency applications, you will use Single mode exclusively and one of three frequency ranges.

Connect a BNC cable to either the 150 kHz or 2 MHz BNC input connector, as appropriate. Both inputs are non-terminating (100 kΩ), so any BNC cable can be used (RG-58, RG-59, RG-174, etc.); for long coax runs, however, low-loss (low capacitance) coax such as RG-58 or RG-59 are preferred. As a result, the CSA-1 can be connected to "bridge" any suitable signal. A BNC "T" connector is ideal for this application. The level applied to the CSA-1 should be in the range of 350 mV (0.35 V) to 3.5 VRMS for the 150 kHz input, and 150 mV (0.15 V) to 2.0 VRMS for the 2 MHz input.

> NOTE: If you use a standard BNC 50 Ω attenuator to reduce a signal applied to the input of the CSA-1, be cautious about the amount of expected attenuation. Those attenuators are often "T" pads (two series resistors, with a shunt resistor at the junction of the two series resistors, to form a network that resembles a "T"), and are designed for a 50 Ω load. Since the CSA-1 is 100 kΩ, insufficient attenuation may result.
>   For example, a 10 dB 50 Ω "T" pad may attenuate the signal by about 4.8 dB, and a 20 dB pad by roughly 14.1 dB, if operated into a high impedance load. Either use a larger attenuator, or add a BNC 50 Ω feed-through terminator after the attenuator.

The fundamental procedure for making measurements is:

1. Select input, Single

2. Select input source

3. Adjust Vertical scale (dB) and Vertical Resolution (VR)

4. Adjust Horizontal Resolution (HR)

5. Select Display Mode

6. Select Window Type

Press MENU SEL and use the MENU ∧ ∨ keys to highlight INPUT SELECT, and then press SEL. Select Single, then choose:

- 150 kHz A/D — Selects the 150 kHz analog BNC input.

- 1 MHz A/D — Selects the 2 MHz analog BNC input, but only displays signals to 1 MHz.

- 2 MHz A/D — Selects the 2 MHz analog BNC input and displays the full 2 MHz bandwidth.

The adjustment of range and resolution and, if desired, the cursors, is basically the same as for audio measurements.

- dB — Press VERT SEL until dB is highlighted, then use the VERT ∧ ∨ keys to select from +100 to –120 dB. Even though the default places 0 at the top of the display, use any value that produces the best on-screen display. Masks, however, are referenced to 0 dB (masks are an extremely powerful tool, and are described below).

- VR — Press VERT SEL until VR (vertical resolution) is highlighted, and use the VERT ∧ ∨ keys to select a resolution of 0.1 dB to 2.5 dB. Since the vertical resolution is 48 points, 2.5 dB allows display of the full 120 dB span, whereas 0.1 dB will display a range of 4.8 dB. There are a total of 7 choices. Depending on your signal source, VRs of 1.5 dB (72 dB), 2.0 (96 dB) and 2.5 (120 dB) produce generally good results.

- C1 — Press VERT SEL until C1 (Cursor 1) is highlighted, and use the VERT ∧ ∨ keys to position the vertical cursor (the horizontal dotted line) to any one of the 48 vertical points. Because of the display's width, it may be difficult to see when a specific frequency exceeds a certain level. Use the vertical cursor as a reference to simplify observations.

- HR — Press HORZ SEL until HR (horizontal resolution) is highlighted, and use the HORZ ∧ ∨ keys to select from a resolution of 10 Hz to 500 Hz in 150 kHz mode; 50 Hz to 5 kHz in 1 MHz mode; and 100 Hz to 10 kHz in the 2 MHz mode.

- C2 — Press HORZ SEL until C2 (Cursor 2) is highlighted, and use the HORZ ∧ ∨ keys to set the position of the horizontal cursor (the vertical dotted line) to any one of 200 horizontal points. The exact level at the cursor location will appear under the frequency (top right-hand corner of the display). Measurements are in dB relative to a reference you choose.

Set the Display Mode. The four options are described below:

- REALTIME — This mode shows the signal in "real-time"; the spectrum display is updated as fast as the FFT is calculated. The update rate varies from 100 milliseconds to 1 second depending on the selected frequency resolution. If the input signal is a continuous tone or unmodulated carrier, the display will show its level and frequency without variation. If the input is a modulated carrier, the display will vary constantly. Use this mode to observe dynamic signal changes.

- PEAK HOLD — This choice continuously "monitors" the signal by continuing to calculate FFTs at the real-time rate for a specified period, then displays the highest level at each frequency during that period. It resets (starts over) at selectable time intervals. Choose this mode to "capture" and display snapshots of peaks of a varying signal, especially if the display is updating too quickly in Realtime mode. The update (reset) rate can be set in 0.1 second increments up to 10 seconds. Unlike Infinite Hold, lower levels can replace higher levels.

- AVERAGE — This option, as its name suggests, "averages" a specified number of FFT samples (from 2 to 64). It is useful for displaying average levels rather than peak, making it less susceptible to random noise "spikes."

- INFINITE HOLD — Similar to Peak Hold, Infinite Hold displays the highest (peak) level calculated in each bin. It accumulates until it is disabled or reset (it does not reset automatically like Peak Hold). A higher level will replace a lower level, but not vice versa. Then use the horizontal cursor (C2) to determine the level in each bin of interest.

The last step is to select the Window type. Discussed above in detail, choose the window that best displays the attributes of the signal you're analyzing. Hamming and the two Blackman-Harris windows are good choices especially if you're working with a mask (these windows minimize leakage into adjacent bins, which is important

in preventing false, out of tolerance indications). The chart below summarizes the characteristics of each Window type.

Window Type	Selectivity	Bin Edge Error	Noise Floor
Rectangular –13	High	High	High
Triangular -27	Medium	Medium	Medium
Hanning –31	Medium	Medium	Medium
Hamming –43	Medium	Medium	Medium
Blackman-Harris –67	Low	Low	Low
Blackman-Harris –92	Very Low	Very Low	Very Low

APPLICATIONS — The adaptability of the Belar CSA-1 in the studio, at the transmitter site and in the field is illustrated in some of the more familiar applications presented below.

Use a dynamic signal source like program material, or a static source like an audio signal generator, a test CD and a CD player (which can be used for a source of digital test signals), or software that can generate test signals.

NOTE: Popular test CDs are: CBS Records Standard Test Disc, CD-1; Denon Audio Technical CD, C39-7147; and the NAB Broadcast and Audio System Test CD, and CD II. The NAB test CDs, in particular, feature a number of useful broadcast-related test signals.

DIRECT EQUIPMENT MEASUREMENTS

■ Observe, in real-time, the level vs. frequency characteristics of an analog composite FM stereo signal from 0 Hz to 150 kHz. Look at the L+R main channel, the 19 kHz pilot and the 38 kHz L-R subcarrier. Determine how much energy appears above 53 kHz as the processor/stereo generator is operated more aggressively. Check for pilot modulation (amplitude variation or "bounce"), and whether the L+R or L-R channels are interfering with the pilot. Connect the CSA-1's 150 kHz input directly to the output of the stereo generator.
    Modulation of the pilot in synchronism with program audio can indicate excessive composite clipping.

■ Monitor the output of an analog composite STL receiver. Compare its output to that measured directly at the output of the stereo generator or input of the STL transmitter. Check for noise caused by interference or path problems. (Mono and dual channel STLs can also be checked using the techniques described in the audio applications section above.)

- Inspect the 57 kHz RDS/RBDS signal directly (use the 150 kHz input). Check the spectrum occupied by the RDS signal to ensure that it's reasonably "clean" and not contaminated by the upper sidebands of the L-R subchannel, the lower sidebands of a subcarrier, or by artifacts from aggressive processing. Estimate its injection using an accurate 19 kHz pilot as a reference.

- Examine a subcarrier directly, regardless of its modulation scheme, again using the 150 kHz input. Confirm that the spectrum it occupies—usually in the range of 60 kHz to 99 kHz—is free from artifacts that can reduce its demodulated signal-to-noise ratio. Use the CSA-1 to help determine the source of any noise or crosstalk. Estimate injection level using an accurate 19 kHz pilot as a reference.

- Analyze the RF output of an AM broadcast transmitter (be careful not to exceed the CSA-1's 2.0 VRMS input level; refer to the note above about using BNC 50 Ω attenuators). Use the 2 MHz input, and select either the 1 or 2 MHz mode. For transmitters below 1000 kHz, inspect their second harmonic. The spectrum from 0 Hz to 1 or 2 MHz, as selected, will be displayed.

- Confirm the calibration of your FM modulation monitor using the carrier-null method. Needed is an accurate, reasonably low-distortion audio generator with level adjustment resolution of less than 0.5 dB, and a Belar DC-4 Frequency Agile Down Converter, or the Down Converter available as Option 2 for the Belar FMMA-1, "The Wizard." The down converter will convert your carrier frequency to a 650 kHz intermediate frequency (IF) signal that will be used for analysis. (Option 2 requires ±15 VDC for operation, which is normally supplied by the FMMA-1. It can, however, be powered by a separate regulated power supply.)

  Connect an RF sample to the down converter, and the down converter to the 2 MHz input of the CSA-1 (the 2 MHz input accepts levels from 150 mV to 2.0 VRMS; use an external attenuator if needed, but heed the note above if a BNC 50 Ω attenuator is used). Configure the CSA-1 as described below:

  1. Select Input, Single, 1 MHz A/D

  2. Select Display Mode, Realtime

  3. Select Window Type, Blackman-Harris –92 dB for the lowest noise floor

  4. Adjust the Vertical scale (dB) to display 0 to –100 dB (actually –120 dB) and set Vertical Resolution (VR) to 2.5 dB

5. Set the Horizontal Cursor (C2) to 650 kHz (this assumes you are using a Belar Down Converter)

6. Change the Horizontal Resolution (HR) to 2.5 kHz, 1 kHz or 500 Hz

Connect the audio generator to the analog composite input of the exciter (disconnect the stereo generator and connect the audio generator directly to the exciter). Next, apply a 31,188 Hz tone (if the generator is not frequency accurate, use a frequency counter to be sure). Finally, increase the level of the tone slowly as you observe the carrier on the CSA-1 (the carrier will appear at the down converter's IF frequency of 650 kHz). To achieve a complete carrier null, you will need level adjustment capability of less than 0.5 dB. When the carrier completely disappears, the transmitter is deviating exactly 75 kHz which, in the U.S., is defined as 100% modulation.

If your generator will not produce 31,188 Hz or you are working with a mono FM exciter, you may employ a 13,587 Hz tone instead. Use the above CSA-1 settings, but change the Horizontal Resolution (HR) to 1 kHz, 500 Hz, or 250 Hz. Connect the generator to the audio input of the exciter (pre-emphasis has no affect on this measurement) and slowly increase the generator's level until the carrier disappears a second time (the first carrier null will occur at approximately 44% modulation, which is about 33 kHz deviation). (The exact frequency of 13,586.8 Hz is available on the NAB Broadcast and Audio System Test CD.)

OFF-AIR MEASUREMENTS

■ For AM off-air measurements, connect the CSA-1 directly to an antenna if sufficient signal strength is available. Use a vertical whip or, preferably, a shielded loop antenna. The Belar LP-1 is a passive (unamplified) shielded loop antenna, and the Belar LP-1A adds an internal amplifier for unity gain operation. Or, if sufficient signal level isn't available, connect the 260 kHz IF output of the Belar RFA-2 AM RF amplifier to the CSA-1. In all cases, use the 2 MHz input of the CSA-1, which allows a clear view of the spectrum from 0 Hz to 1 or 2 MHz, as desired. (In the case of the RFA-2 RF amplifier, the carrier of interest will appear at 260 kHz rather than the actual carrier frequency.) (Note: The RFA-2 is 0.2 dB ±10 kHz; –3 dB ±16 kHz; and greater than –40 dB ±40 kHz.)

Shielded loops are preferred because they reject electro-mechanical interference (EMI) and off-axis interfering stations. The unamplified LP-1 has an attenuation of 30 dB while the amplified LP-1A is non-attenuating due to the gain of its internal amplifier. Sometimes, even in high signal areas, the LP-1A is recommended because the amplifier isolates long coax runs from the antenna loop. In that case, however, it may be desirable to convert the amplifier to unity gain operation (instead of 30 dB gain) to prevent overloading an input. (The LP-1A requires 12 to 15 VDC,

which is normally supplied by the Belar RFA-2 or Option 1, an optional power supply.)

- A shielded loop antenna, such as the Belar LP-1 or LP-1A, described above, can be used for AM NRSC "field" measurements. Connect the antenna to the 2 MHz input of the CSA-1 and use the factory programmed NRSC mask (masks are described below). The CSA-1 and Option 1 (the power supply for the Belar LP-1A active shielded loop antenna), if used, can be powered by a commonly available automotive inverter (12 VDC to 120 VAC), or a UPS (uninterruptible power supply) frequently used with computer systems.

- For FM off-air carrier measurements, use a high-quality RF amplifier that provides an IF output such as the Belar RFA-1A fixed frequency, or the RFA-4 frequency-agile FM RF amplifier. Similar to FM transmitter measurements above, connect the 650 kHz output of the RF amplifier to the 2 MHz input of the CSA-1. The spectrum from 0 Hz to 1 or 2 MHz (as selected) will be displayed, but remember that the carrier frequency, determined by the RF amplifier, will appear at 650 kHz on the CSA-1 (in the RFA-1A and RFA-4, the first adjacent channel is attenuated by 27 dB, and the third adjacent by 46 dB).

- Inspect the off-air demodulated spectrum of an FM transmitter from 0 Hz to 150 kHz by using an RF amplifier, such as the Belar RFA-1A fixed frequency, or the RFA-4 frequency-agile version, and a high-precision FM demodulator such as the Belar FMMA-1, "The Wizard." (For accurate, meaningful measurements, it is imperative that you use a high-quality RF amplifier and FM demodulator.) Connect the composite (wideband, non-de-emphasized) output to the 150 kHz input of the CSA-1. Like the direct equipment measurements above, view the entire spectrum from 0 Hz to 150 kHz, or zoom in on any portion thereof. You'll see the L+R main channel, the 19 kHz pilot, 38 kHz L-R subchannel and, if broadcast, the RDS/RBDS carrier at 57 kHz, and any subcarriers.
  Look for modulation of the 19 kHz pilot, which is an indication of excessive composite clipping or multipath. Approximate injection levels of the RDS and subcarriers can be checked using an accurately calibrated 19 kHz pilot as a reference. Also, carefully examine the spectrum in the upper region, approaching 150 kHz. First adjacent channel interference, if any, will be clearly visible.

Display Storage

The CSA-1's storage function is a convenient method for either saving measured spectrum curves for later reference, or comparing previously measured curves to new measurements. Up to 10 different curves can be stored in non-volatile memory. To store a displayed curve, choose Menu, Storage, Store, then a number from 0 to

9. Capture a curve "on the fly"—if the Display Mode is set to Realtime or Average—or store curves acquired using either Peak Hold or Infinite Hold.

Stored curves can be displayed by accessing Menu, Storage, Recall, then selecting the appropriate curve number (0 to 9). You may either display the curve (Menu, Storage, Display, On) or subtract it and display the difference from an on-screen curve (Menu, Storage, Subtract, On). A brief example of how to use the Storage feature is described below.

Configure the CSA-1 to display, say, a Single L+R curve using the 24 kHz L+R A/D or AES/EBU input. Adjust the vertical and horizontal resolutions as desired, and set the Display Mode to Infinite Hold. Then play a portion of audio (we used a 128 Kbps MP3 file) and store it as 0. Clear the display (Menu, Display Mode, Realtime, then Menu, Display Mode, Infinite), and play a different section of audio (we used the source WAV file). Finally, display the stored curve (Menu, Storage, Recall, 0, then Menu Storage, Display, On). As you can see in the figure below, the differences between the source WAV file (curve filled) and the stored MP3 file (curve only) are readily apparent. (Yes, there is an absence of high frequencies in the MP3 file.)

(Insert screen shot here.)

Spectral Masks

Masks are yet another useful feature of the Belar CSA-1. A spectral mask is a set of "limit" lines—the lines can be horizontal, vertical or diagonal—that define the desirable or mandatory amplitude vs. frequency characteristics of a signal. These lines are overlaid on the display, like cursors, to serve as visual boundaries. Up to 8 user-defined masks can be stored in non-volatile EEPROM, and 2 are preprogrammed: FM (IBOC) and AM NRSC-2.

Masks are referenced to 0 dB to allow direct reading of measurements. Therefore, any signal being measured must also be calibrated to 0 dB. This is accomplished by using the rear panel trim pot associated with the input being used (these are 10-turn trim pots, so precise adjustment is usually quite easy). The CSA-1 accommodates a wide input voltage range so amplification should seldom be required; usually high level signals only will require attenuation (refer to the note above about using BNC 50 Ω attenuators).

Fine adjustments can be made using Menu, Vertical Axis, 0 dB Ref Level. Normally, this reference is set to –3.0 dBFS, which allows for 3 dB of "headroom" above 0 dB. Consequently, signals that exceed 0 dB on-scale will not be clipped until they reach +3 dB. Use this option to make fine adjustments, on the order of 1 dB or less; otherwise, use the rear panel trim pot (remember to reset it to the recommended –3.0 dB at the conclusion of your measurements). Setting this reference significantly below –3.0 dBFS will result in a corresponding reduction in dynamic range (the noise floor will effectively rise) thus limiting the instrument's ability to display low-level signals.

To display a mask:

1. Select Masks, Display, On

2. Select Masks, Recall, FM (IBOC) for example

3. Select Horizontal Axis, Freq Scale, +/- Offset

4. Select Horizontal Axis, Center Frequency, and change the default from 500 kHz to 650 kHz (if you're using a Belar FM Down Converter or RF Amplifier)

5. Adjust Vertical scale (dB) and Vertical Resolution (VR)

6. Adjust Horizontal Resolution (HR)

7. Select Display Mode (use Peak Hold to view snapshots of the signal, or Infinite Hold to capture the peaks over time)

8. Select Window Type (as discussed above, Hamming and the two Blackman-Harris windows are preferred in this application because these window types minimize leakage into adjacent bins, which is important in preventing false indications that might suggest out-of-tolerance operation)

Although you can use a mask without switching to Center Frequency mode (steps 3 and 4 above), doing so keeps the mask and signal perfectly centered at all horizontal resolution settings.

Another valuable feature is the Subtract function. Accessed via Masks, Subtract, the selected mask will be subtracted from the signal, and the difference will be displayed. This is especially useful for highlighting minor differences between a mask and a measured signal.

Documenting the Measurements

Measurements made on the CSA-1 can either be saved to disk or printed. Both methods rely upon the supplied Windows™-based software.

Install the software on a compatible PC, and connect its serial port to the RS-232 port on the CSA-1 (a null modem is not required).

Along with full control of the CSA-1, the software allows you to print measured data or save it to disk.
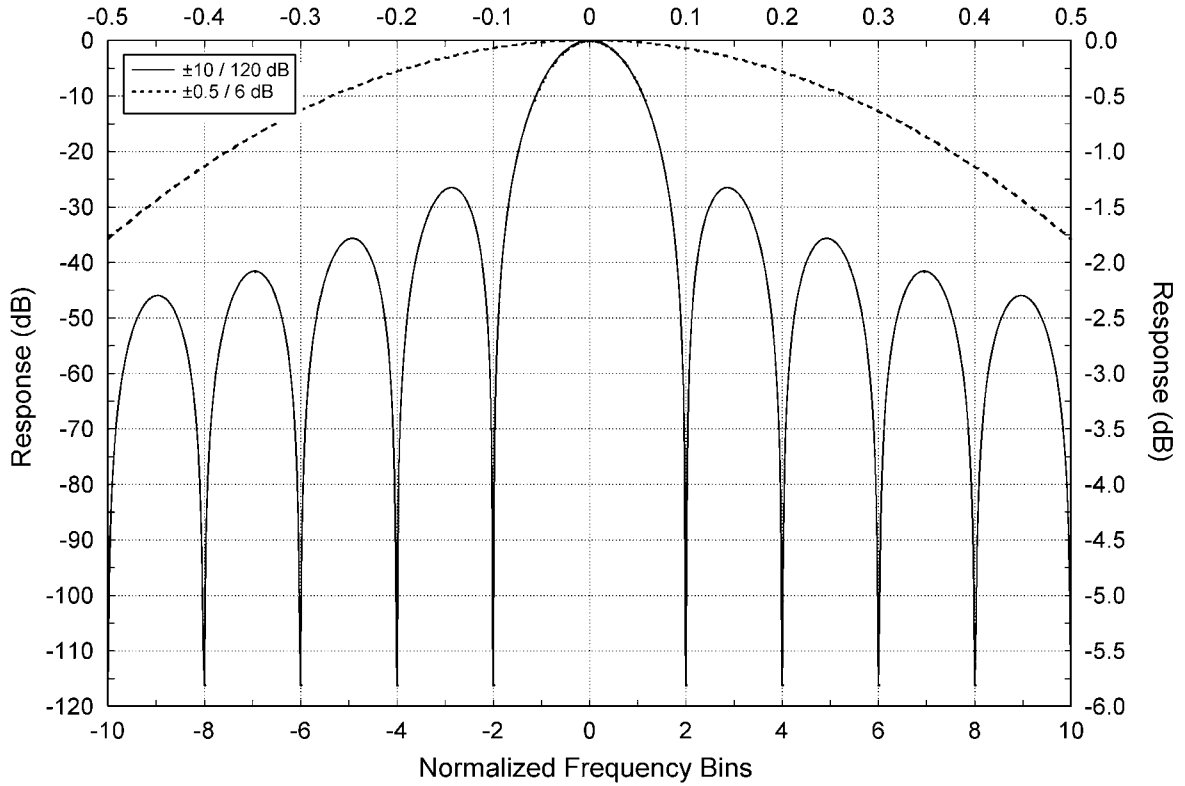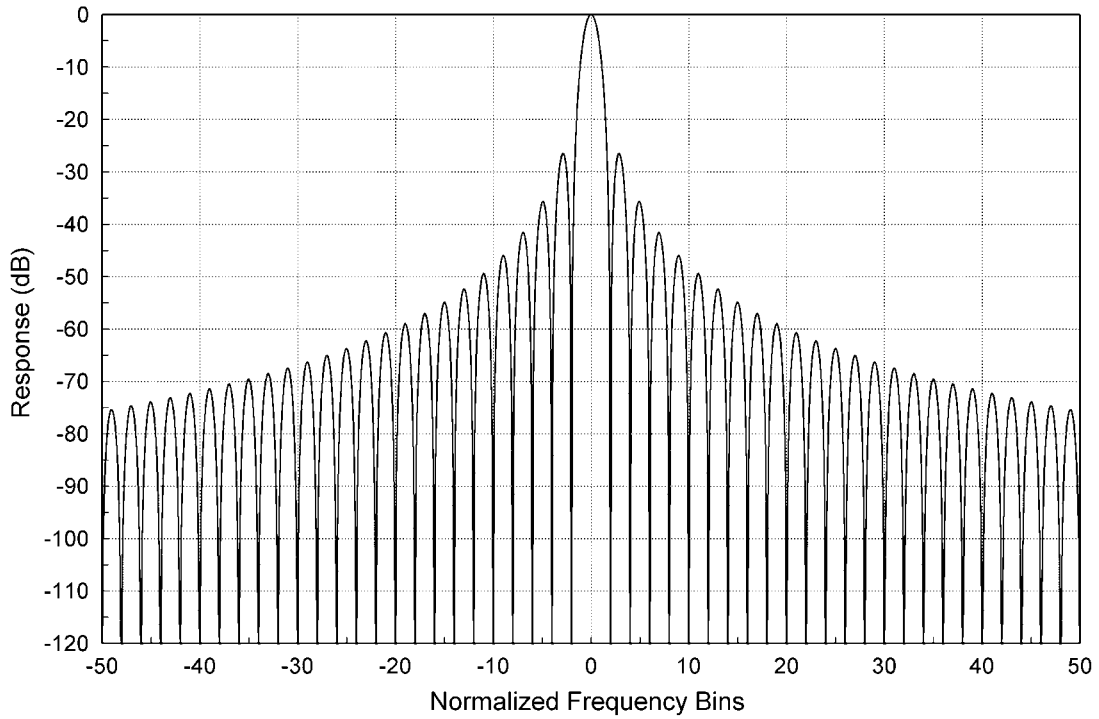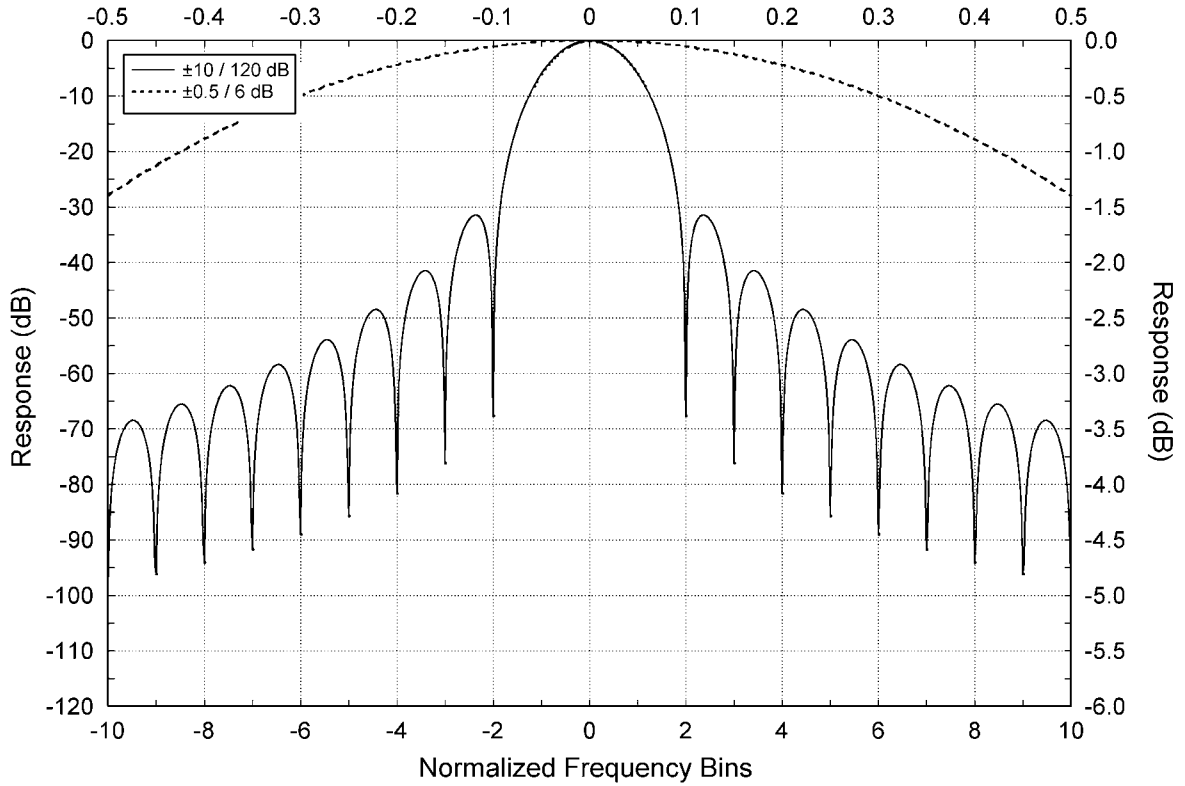
Rectangular -13 dB
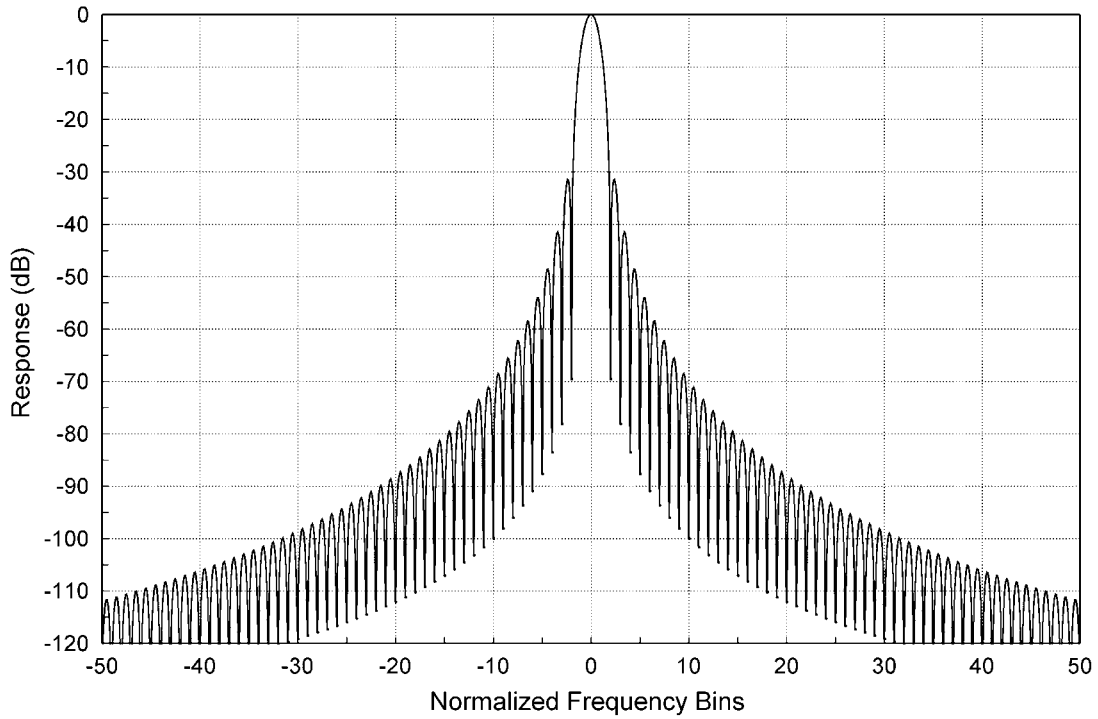
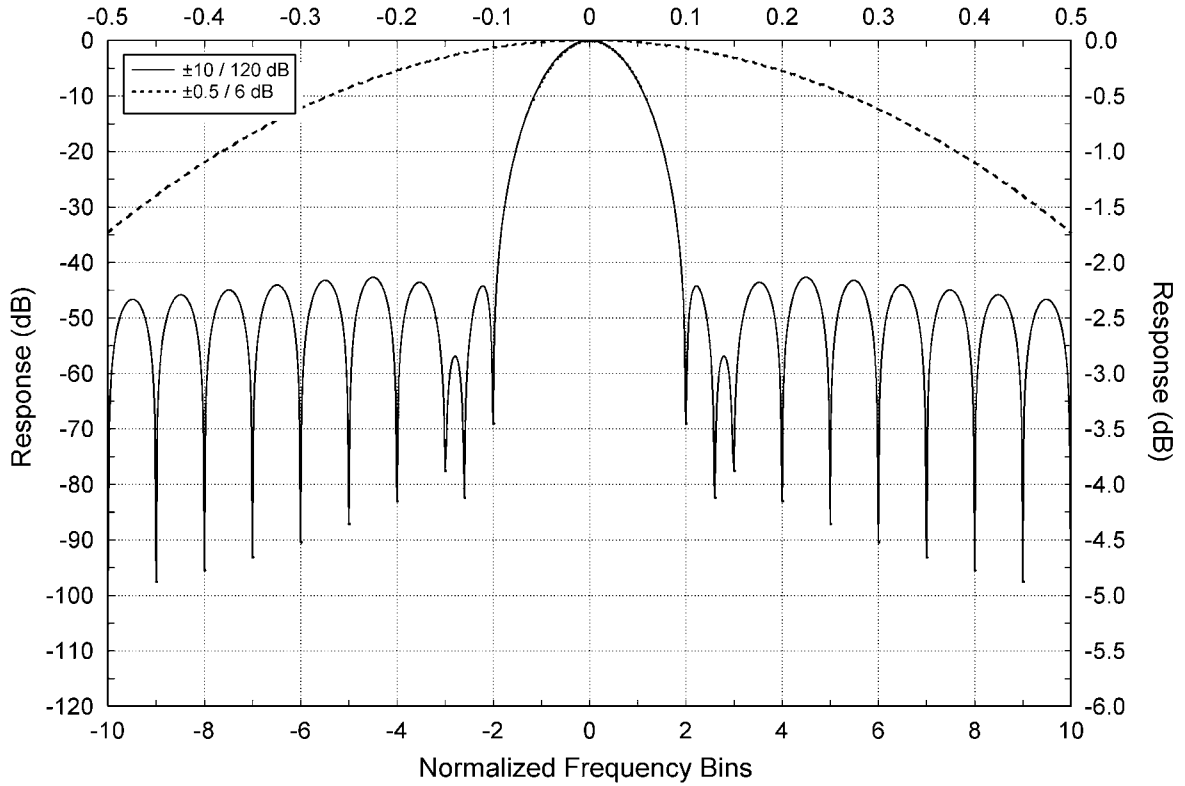Rectangular -13 dB

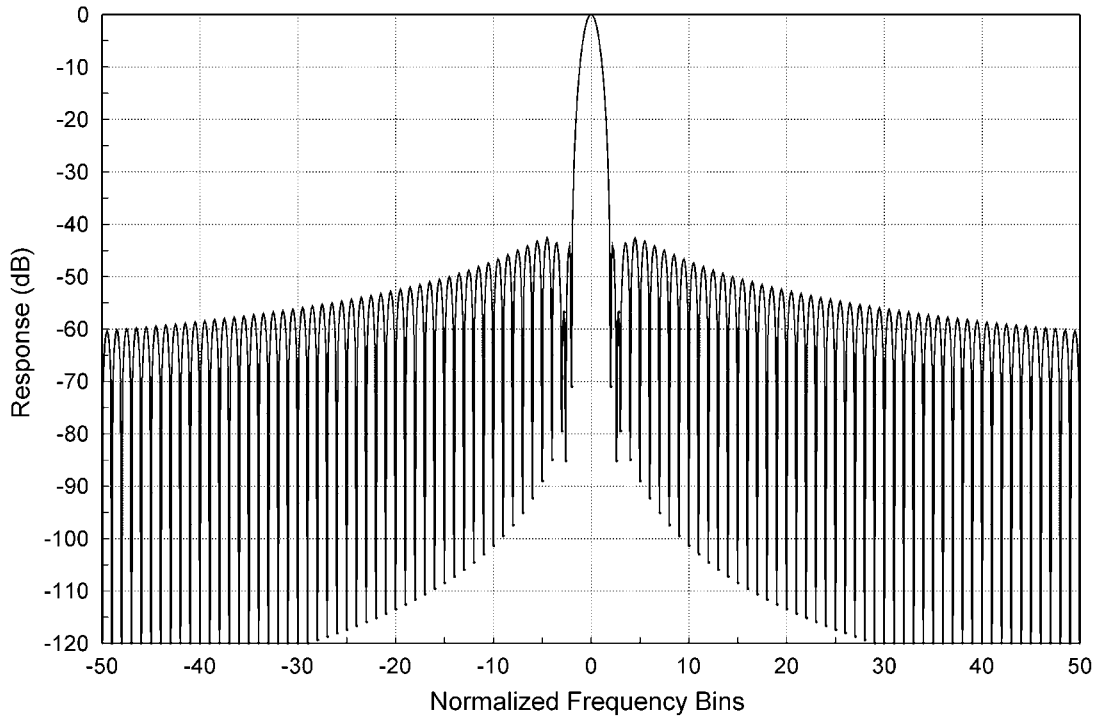# Triangular -27 dB
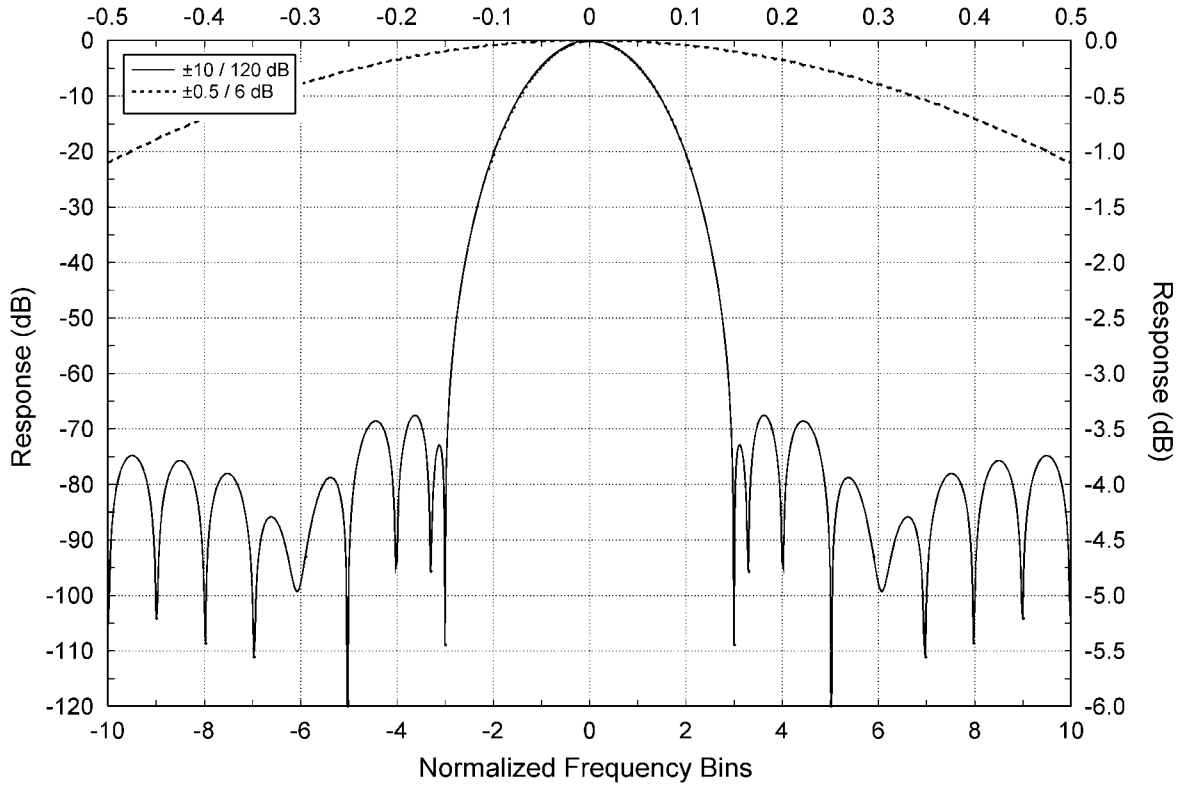

Triangular -27 dB

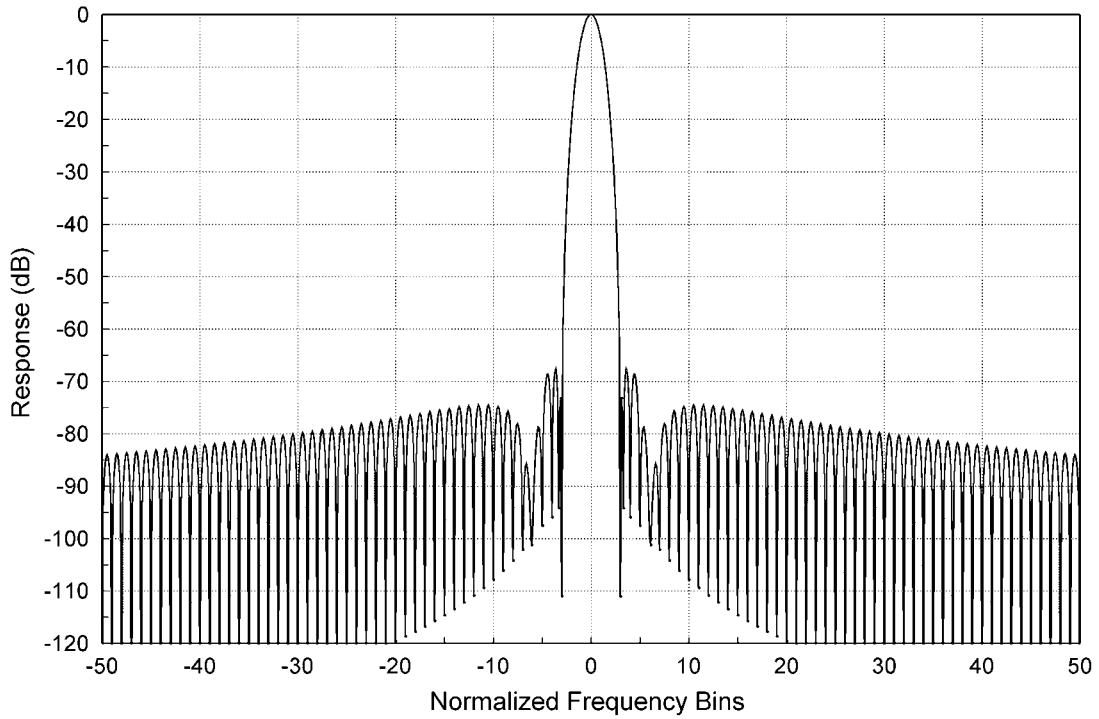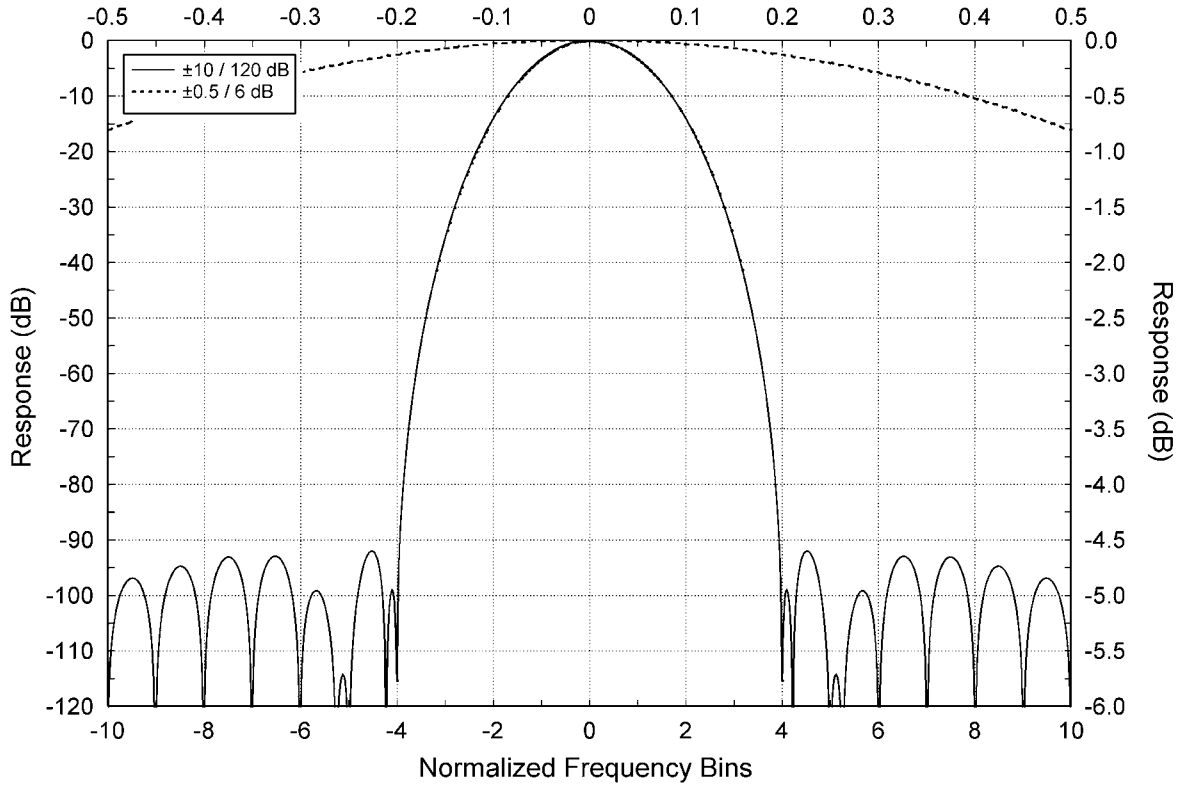# Hanning -31 dB

Hanning -31 dB
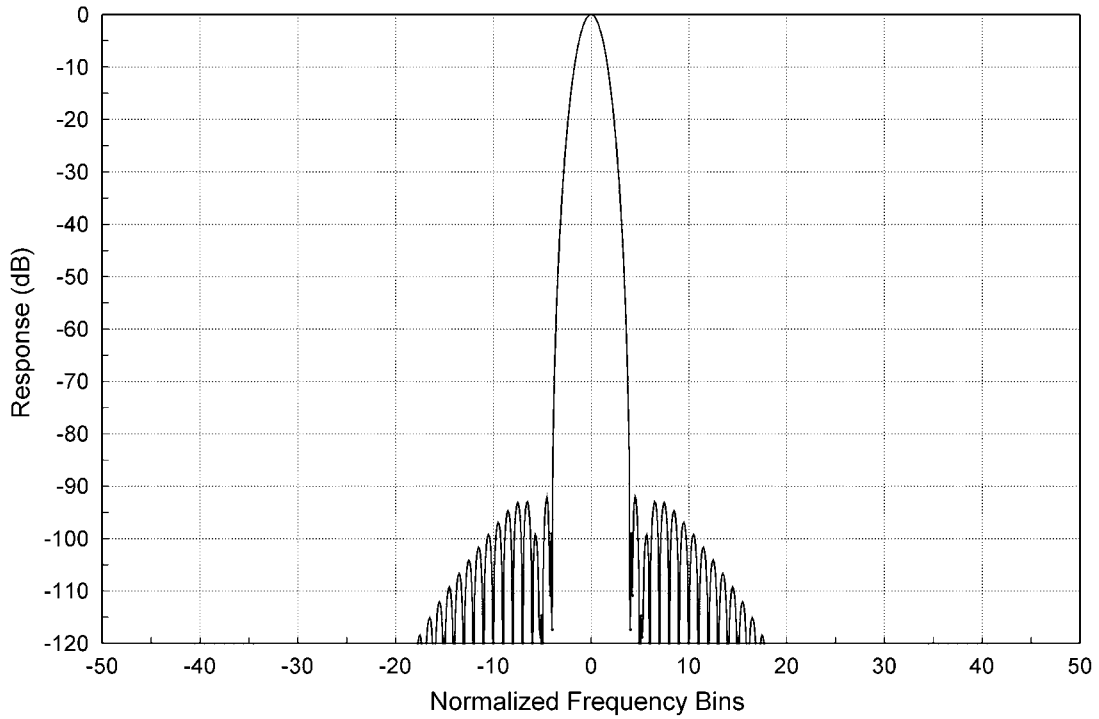
Hamming -43 dB

Hamming -43 dB

Blackman-Harris -67 dB

Legend:
— ±10 / 120 dB
···· ±0.5 / 6 dB

Response (dB) (left axis, top plot: 0 to -120)
Response (dB) (right axis, top plot: 0.0 to -6.0)
Normalized Frequency Bins (bottom axis: -10 to 10; top axis: -0.5 to 0.5)

Blackman-Harris -67 dB

Response (dB) (0 to -120)
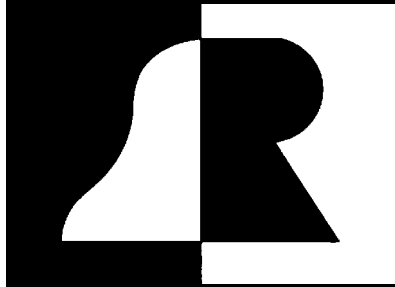Normalized Frequency Bins (-50 to 50)

Blackman-Harris -92 dB

Blackman-Harris -92 dB

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