Application Note 6

Digital Audio Broadcast Receiver Audio Testing

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Digital Audio Broadcast (DAB) Receiver Audio Testing

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Objective

The objective of this application note is to describe accurate, timesaving techniques for testing the audio quality of Digital Audio Broadcast (DAB) receivers with Audio Precision 2700 series and ATS-2 audio analyzers.

While the information in this document focuses on DAB as it applies to satellite radio transmissions and reception, and more specifically to appropriate receiver testing techniques, there is also useful information contained here applicable to the larger universe of DAB technology.

The problem most frequently encountered in the audio testing of DAB receivers relates to signal mute intervals (dropouts) caused by the receiver when stimulated with a synthesized radio signal from a radio frequency (RF) generator. In order to avoid unreliable measurements during these dropouts, it is critical that audio tests be performed in synchronization with your RF generator.

The testing techniques described here—addressing such solutions as external triggered measurements, audio mute detection, and multitone signals—will solve typical dropout problems, resulting in improved measurement accuracy and repeatability, and decreased measurement time.

The audience that will benefit most from this information will be DAB receiver design engineers and DAB broadcast design engineers. However, manufacturing test engineers developing production test QA procedures will also find this document valuable for its techniques that effectively reduce receiver test time.

The CD-ROM accompanying this application note contains test files and macros that implement these testing techniques.

Introduction to Digital Audio Broadcasting

Digital Audio Broadcasting, or DAB, is the radio frequency transmission of one or more audio signals with the apparent quality of a Compact Disc. These transmissions are achieved with reduced bit rate coding by a Perceptual Audio Codec (PAC) prior to transmission, and decoding with the complementary PAC decoder at the receiver.

THE DAB BROADCASTING SYSTEM

Generation of ETSI DAB Signals

Figure 1 shows how an ETSI (Eureka 147) DAB service signal is coded individually at the source level through a PAC in order to reduce the bit rate, and then error-protected and time-interleaved in a channel coder.

These service signals are multiplexed in the Main Service Channel (MSC), according to a pre-determined multiplex configuration. The multiplexer output is combined with Multiplex Control and Service information to form transmission frames in the Transmission Multiplexer. Orthogonal Frequency Division Multiplexing (OFDM) is then applied to shape the DAB radio frequency signal, which consists of a large number of carriers. Coded Orthogonal Frequency Division Multiplexing (COFDM) is used for some DAB systems. For more information about OFDM and COFDM see web tutorials at www.palowireless.com. The signal is then transposed to the appropriate radio frequency band, amplified, and transmitted.

Reception of ETSI DAB Signals

DAB receivers consist of a tuner, demodulator, channel decoder, audio decoder, and audio output. Figure 2 demonstrates a conceptual DAB receiver. The DAB signal is selected in the analog tuner, the digitized output of which is fed to the OFDM demodulator and channel decoder. The information contained in the Fast Information Channel (FIC) is passed to the user interface for service selection and is used to setup the receiver appropriately. The Main Service Channel (MSC) data is further processed in a perceptual audio decoder to produce the left and right audio signals (analog and/or digital).

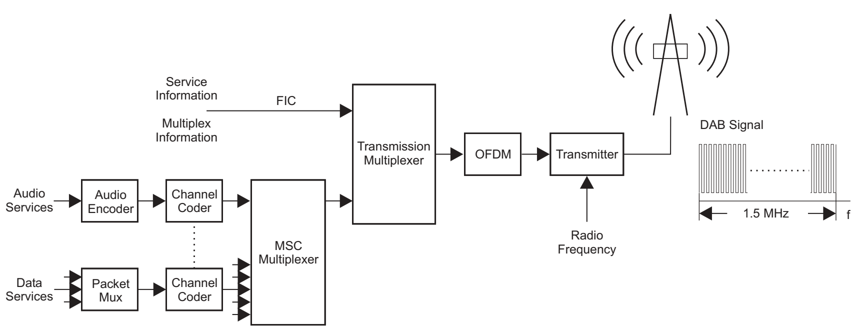


Figure 1. Generation of the DAB Signal (courtesy Eureka 147 consortium).

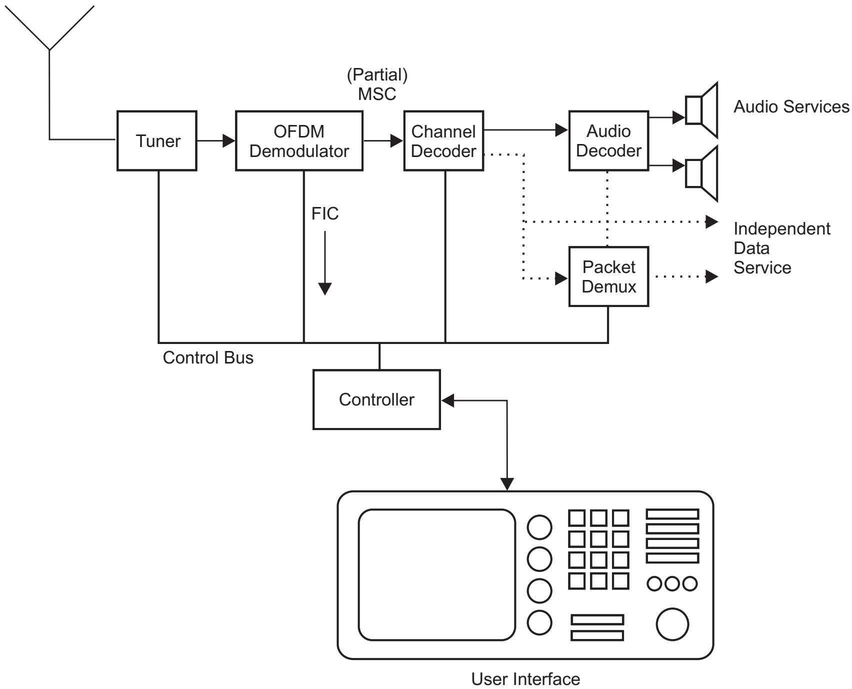


Figure 2. Conceptual DAB Receiver (courtesy Eureka 147 consortium).

Analog audio is generated with conventional digital/analog (D/A) converters and output circuitry. Additional receiver technologies manage multiple program streams, data streams, auxiliary services, and error correction. Broadcasting is achieved through satellite and terrestrial transmitters to receivers in automotive entertainment systems, portable radios, and desktop entertainment systems in homes and businesses. Some DAB broadcast services may be provided at no cost to listeners (as it is in the U.K.) or may be purchased by subscribing to a service (as is the case in the U.S.). DAB receivers, unveiled commercially in the United States in 2000, are available from many manufacturers, and are generally referred to as “mobile,” “portable,” or “fixed.”

Terrestrial Broadcasting

Terrestrial Digital Audio Broadcasting relies on signal transmission from ground antennas.

The US Federal Communications Commission (FCC) has selected HD Radio (In-Band On-Channel system technology from iBiquity Digital) as the terrestrial DAB system in the United States. HD Radio technology accommodates an easy transition for AM and FM broadcasters and consumers by using the existing radio infrastructure and spectrum while preserving the existing analog service for as long as is necessary.

The World Wide DAB Forum promotes the digital radio system developed by the Eureka 147 Project, named after a technical consortium that developed the European Telecom Standards Institute (ETSI) standards for DAB (see http://pda.etsi.org/pda/queryform.asp and query for “DAB”). This format has been implemented in Europe and its use is growing throughout the world.

Satellite Digital Audio Radio Services (SDARS)

Satellite Digital Audio Radio Services (SDARS) involve the transmission of a large number of simultaneous audio streams from orbiting satellites, and subsequently through terrestrial repeaters, to compatible mobile, portable, and fixed receivers.

In the United States, the FCC has approved two, competing SDARS companies, both currently operating (as of 2003) in North America: Sirius Satellite Radio and XM Radio. These companies’ SDARS signals are transmitted on the S-band between 2320.0 MHz and 2345.0 Mhz. Each company is allotted a bandwidth of 12.5 MHz. Currently, both companies service more than 100 channels of continuous digital audio. The SDARS companies are in direct competition with the more than 12,000 conventional U.S. radio stations, which currently broadcast to an estimated 500 million radio receivers.

Sirius Satellite Radio broadcasts to Sirius-compatible receivers from three satellites supported by terrestrial repeater stations. These satellites are located in high-elevation / high-angle geosynchronous elliptical orbits over the North American continent, and appear to move in a figure eight pattern.

XM Radio broadcasts to XM Radio compatible receivers from two geosynchronous satellites at fixed sky positions over the equator. High power transmitters cover the continent from a lower southern angle. Terrestrial repeater stations support these satellites.

The Global Radio Satellite System plans a similar system in Europe. The proposed system will be based on the ETSI DAB standard but using an MPEG-4 AAC codec for higher quality audio at low bit rates (compared to MPEG-2 Layer 2). Current ETSI DAB receivers are incompatible with the proposed system.

In addition to multipath interference issues, satellite radio systems must deal with greater reception problems than conventional FM radio does. To compensate for this problem, a network of terrestrial transmitter stations re-broadcast signals into “shadow” areas shielded from direct satellite signals (such as receivers located behind mountains or tall buildings in urban areas).

Does a PAC degrade audio test signals?

A perceptual audio codec (PAC) in a DAB system reduces the bit rate required to transmit digital audio signals by taking advantage of the psychoacoustic masking of the human auditory system; placing quantization noise close to high-level signals where the noise can't be heard does this. The question often arises whether or not such a codec distorts a single-tone audio test signal sufficiently to make it unusable for distortion testing of a DAB receiver analog output stage.

The ETSI DAB standard specifies an MPEG-2 codec, but proprietary low bit rate codecs are used in SDARS systems. Do these codecs produce a level of quantization noise that would prohibit meaningful distortion measurements? The answer lies in the nature of the test signal.

The behavior of all PACs is similar in how they affect a single-tone sine wave signal. The sine wave signal is passed through the coding and decoding process with a relatively low level of distortion as measured by a traditional notch filter distortion analyzer. This relatively low distortion behavior of a PAC makes it possible to use a transmitted sine wave tone for receiver audio quality testing of the D/A converters and analog output circuitry.

Figure 3 illustrates that a low bit-rate PAC can produce a very clean single tone test signal. The FFT graph shows a 1 kHz sine wave at –60 dBFS level, measured at the digital output of the receiver. This signal drives a stereo D/A converter to produce analog output signals. In this particular receiver, the highest noise or harmonic component is below –110 dBFS, about 50 dB below the fundamental. The digital THD+N distortion measurement with the Audio Precision DSP Audio Analyzer was about –31 dB for both channels. When analog output distortion was measured, the D/A converter in this receiver exhibited a dynamic range of –89 dB (THD+N distortion of –29 dB with a –60 dBFS input signal). This is performance on par with a 16 bit D/A converter in a consumer-grade CD player.

The conclusion is that a perceptual audio codec does not significantly degrade a single-tone test signal.

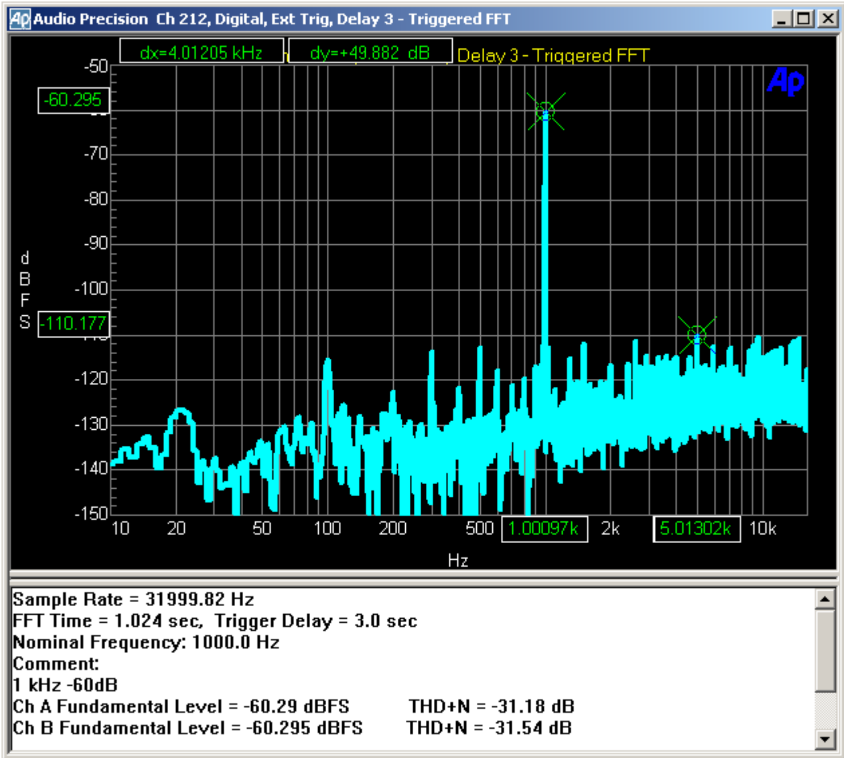


Figure 3. Audio Precision 2700 series instrument using the Spectrum Analyzer tool. The DAB system Perceptual Audio Codec is capable of passing a –60 dBFS single tone sine wave test signal without producing an objectionable level of noise or distortion.

However, more complex test signals such as multitones exhibit increased noise around the frequencies of the test tones due to the nature of the PAC processing algorithms. As the number of sine wave tones increases, more quantization noise is produced around the frequency region of each tone. This prohibits use of a densely populated multitone for distortion test purposes. A multitone processed by a PAC may be used for level, frequency response, and cross-talk tests, but is not usable for distortion testing.

DAB Receiver Audio Test Systems

A DAB receiver audio test system consists of a DAB receiver (and power supply in the automotive world), RF signal generator capable of generating a complex multiple channel DAB radio signal, antenna coupler, and an audio analyzer with digital and analog inputs. This limited system is adequate for QA testing on a production line.

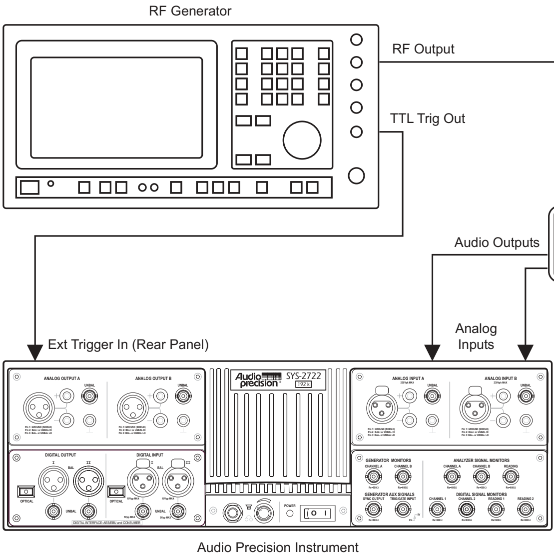


Figure 4. DAB Receiver Audio Test System .

Additional equipment is required for design verification tests or type acceptance tests intended to verify performance against design limits. A type acceptance test system consists of the same audio test equipment used for production QA testing. However, it includes additional RF generators and antenna couplers to simulate radio frequency multipath and interference signal sources, and to simulate delayed signals from terrestrial repeater stations.

Both test systems require an audio analyzer capable of measuring sine wave tones and more complex signals such as: shaped noise, two-tone inter-modulation signals, and multitones. The Audio Precision 2700 series and the ATS-2 are both capable of making these measurements on DAB receivers with analog and digital outputs.

The RF generator must provide a genuine DAB radio signal comprised of multiple channels, each of which carries a different audio test signal. The RF generator has a maximum memory size for digital modulation data that limits the duration of the audio test signal within each channel. For a 100+ channel SDARS signal, this memory must be divided among the channels. Consequently, the audio test signal in each channel will be present as a repeating loop of limited duration, typically 10 to 20 seconds. At the end point of each loop, the receiver mutes the channel output for up to 1 second to silence any discontinuity. Audio measurements must not be performed during the mute interval. To prevent this, it is necessary to carefully synchronize the acquisition of audio measurements with the RF generator.

SDARS Receiver Audio Test Sequence

Due to the limited-duration digital modulation data available in an RF generator, test signals must be distributed in multiple audio program channels. Over 100 program channels can be produced simultaneously for SDARS receiver testing, although a smaller number is used in practice.

An example of an audio test channel plan is shown in the table below:

Channel #	Signal Description	Frequency	Left Level	Right Level	Trigger Delay
1	Sine wave 1 kHz L & R Reference	1 kHz	–3 dBFS	–3 dBFS	2
2	Sine wave 20 Hz L & R	20 Hz	–3 dBFS	–3 dBFS	1
3	Sine wave 100 Hz L & R	100 Hz	–3 dBFS	–3 dBFS	1
4	Sine wave 7.5 kHz L & R	7.5 kHz	–3 dBFS	–3 dBFS	3
5	Sine wave 10 kHz L & R	10 kHz	–3 dBFS	–3 dBFS	8
6	Sine wave 14 kHz L & R	14 kHz	–3 dBFS	–3 dBFS	3
7	Sine wave 15 kHz L & R	15 kHz	–3 dBFS	–3 dBFS	3
8	Sine wave 1 kHz L Crosstalk	1 kHz	–3 dBFS	–Inf dBFS	2
9	Sine wave 1 kHz R Crosstalk	1 kHz	–Inf dBFS	–3 dBFS	2
10	Sine wave 20 Hz L Crosstalk	20 Hz	–3 dBFS	–Inf dBFS	1
11	Sine wave 20 Hz R Crosstalk	20 Hz	–Inf dBFS	–3 dBFS	1
12	Sine wave 100 Hz L Crosstalk	100 Hz	–3 dBFS	–Inf dBFS	2
13	Sine wave 100 Hz R Crosstalk	100 Hz	–Inf dBFS	–3 dBFS	2
14	Sine wave 7.5 kHz L Crosstalk	7.5 kHz	–3 dBFS	–Inf dBFS	2
15	Sine wave 7.5 kHz R Crosstalk	7.5 kHz	–Inf dBFS	–3 dBFS	2
16	Sine wave 14 kHz L Crosstalk	14 kHz	–3 dBFS	–Inf dBFS	3
17	Sine wave 14 kHz R Crosstalk	14 kHz	–Inf dBFS	–3 dBFS	3
18	Sine wave 15 kHz L Crosstalk	15 kHz	–3 dBFS	–Inf dBFS	3
19	Sine wave 15 kHz R Crosstalk	15 kHz	–Inf dBFS	–3 dBFS	3
20	Sine wave 1 kHz L & R D/A Dynamic Range	1 kHz	–60 dBFS	–60 dBFS	2

An audio testing sequence involves selecting a program channel on the receiver (manually or automated) followed by audio measurements appropriate to the type of signal on the channel. Measurements typically consist of level, stereo level balance, THD+N Ratio, crosstalk, and noise. A THD+N ratio measurement is used to test D/A converter dynamic range with a –60 dBFS sine wave test tone (test channel 20). Testing all 20 channels takes a significant amount of time. The tuner takes time to select a new program channel and measurements must be synchronized in time to avoid the mute interval somewhere in the 10 second repeat loop of the test signal. A faster method is needed.

A multitone test signal provides that faster method. The multitone signals in the chart below provide all the frequencies required to test level accuracy, frequency response, and crosstalk for a receiver with a 16 kHz bandwidth in an SDARS system using a 32 kHz audio sample rate. The table below adds these multitone channels to the test channel plan.

Channel #	Signal Description	Frequency	Left Level	Right Level	Trigger Delay
21	Multitone Stereo Waveform File: 32kMulTonIso3perOct.AGM in both channels 32 kHz sample rate Crest Factor 3.975	19.53 25.39 31.25 39.06 50.78 64.45 80.08 101.56 126.95 160.16 201.17 253.91 320.31 402.34 507.81 640.62 806.64 1000.0 1259.77 1587.89 2000.0 2519.53 3175.78 4000.0 5039.06 6349.61 8000.0 10080.08 12699.22 15000.0 15998.05	–24.89 dBFS –24.89 dBFS	–24.89 dBFS –24.89 dBFS	3
22	Multitone Left Waveform File: 32kMulTonIso3perOct.AGM in left channel only 32 kHz sample rate Crest Factor 3.975	31 tones 19.53 Hz–15998.05 Hz	–24.89 dBFS	–Inf dBFS	3
23	Multitone Right Waveform File: 32kMulTonIso3perOct.AGM in right channel only 32 kHz sample rate Crest Factor 3.975	31 tones 19.53 Hz–15998.05 Hz	–Inf dBFS	–24.89 dBFS	3

A more efficient test sequence for production QA testing uses multitone test channels for testing level, response, and crosstalk (channels 22 and 23) and single tone test channels for THD+N distortion and A/D dynamic range (channels 1 and 20), a sequence of only four program channels. This is much faster than sequencing through channels 1 through 20.

The multitone waveform described above has 31 frequencies spaced at third-octave frequencies, providing a comprehensive test. Test time is not affected by the number of frequencies in the multitone because only one measurement acquisition is required per test channel. Both stereo channels of the receiver are acquired and measured simultaneously.

Use the Multitone Creation Utility included with the 2700 series or ATS- 2 to develop a multitone signal to suite your particular receiver test requirements. The topic of multitone testing will be covered in more detail below.

Triggering with an External Trigger Signal

DAB receiver audio testing requires triggering audio measurements in synchronization with the mute interval in a particular receiver channel in order to avoid making measurements during the mute. If the RF generator provides a synchronizing trigger pulse then the Audio Precision analyzer may use this external trigger to determine when to acquire measurements.

The mute interval may not always occur at the same point in time relative to the trigger from the RF generator, so a unique time delay is required for each channel. This may be accomplished by synchronizing and delaying measurements with a trigger signal provided by the RF generator (if a trigger signal is available). The RF generator may provide a TTL logic trigger pulse for this purpose.

Figure 5 shows waveforms of the external trigger pulse from the RF generator and one channel of analog output from a DAB receiver. The waveforms were captured on an Audio Precision 2700 series instrument. The audio mute starts about 7.9 seconds after the rising edge of the trigger pulse and continues until 8.35 seconds. Reliable audio measurements could be acquired immediately after the external trigger for another 8 seconds, or could be delayed until after the mute interval, perhaps 8.5 seconds after the trigger.

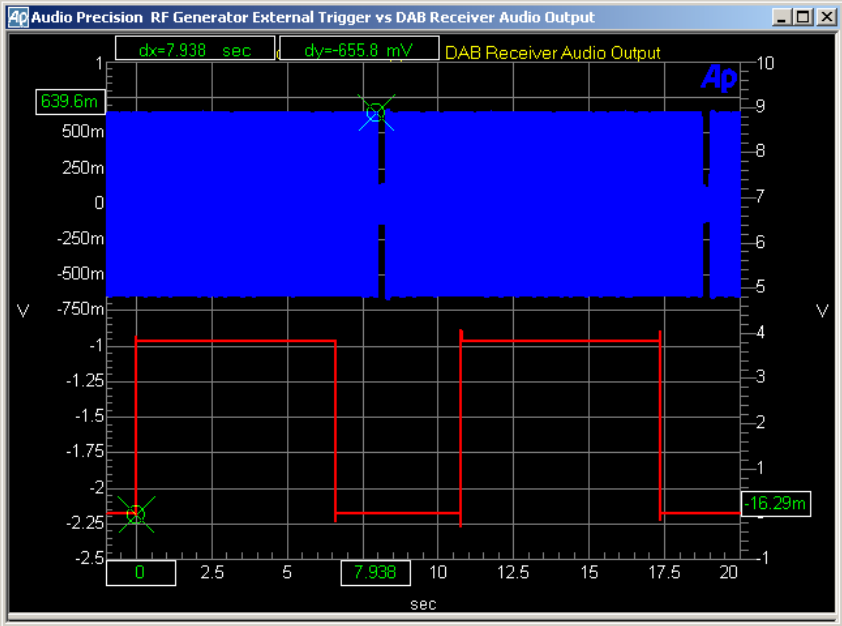


Figure 5. 2700 series Spectrum Analyzer display. The external trigger pulse occurs 8 seconds before the end of the audio mute interval in this DAB receiver program channel.

Triggering audio measurements can be achieved by using the EXT TRIGGER IN connector on the rear panel of the 2700 series dual domain instrument, or the TRIG IN connector on an ATS-2. This external trigger input signal may be used to trigger an acquisition with the Spectrum Analyzer or the Multitone Analyzer but is not usable with the DSP Audio Analyzer.

In order to use the DSP Audio Analyzer for triggered measurements of level, frequency, THD+N, and crosstalk it is necessary to use an AP Basic macro to implement an external trigger sequence. Figure 6 illustrates the external trigger test sequence of the Spectrum Analyzer followed by the DSP Audio Analyzer. The sequence requires the Spectrum Analyzer to trigger on the external event, then load the DSP Audio Analyzer and acquire measurements.

Timing is important. DSP Audio Analyzer measurements need to be completed as soon as possible after the external trigger event. Stereo DSP Audio Analyzer measurements of level and THD+N may be acquired in under 4 seconds. This measurement time is not a problem because the duration of the test signal after the mute is typically 10 seconds or more.

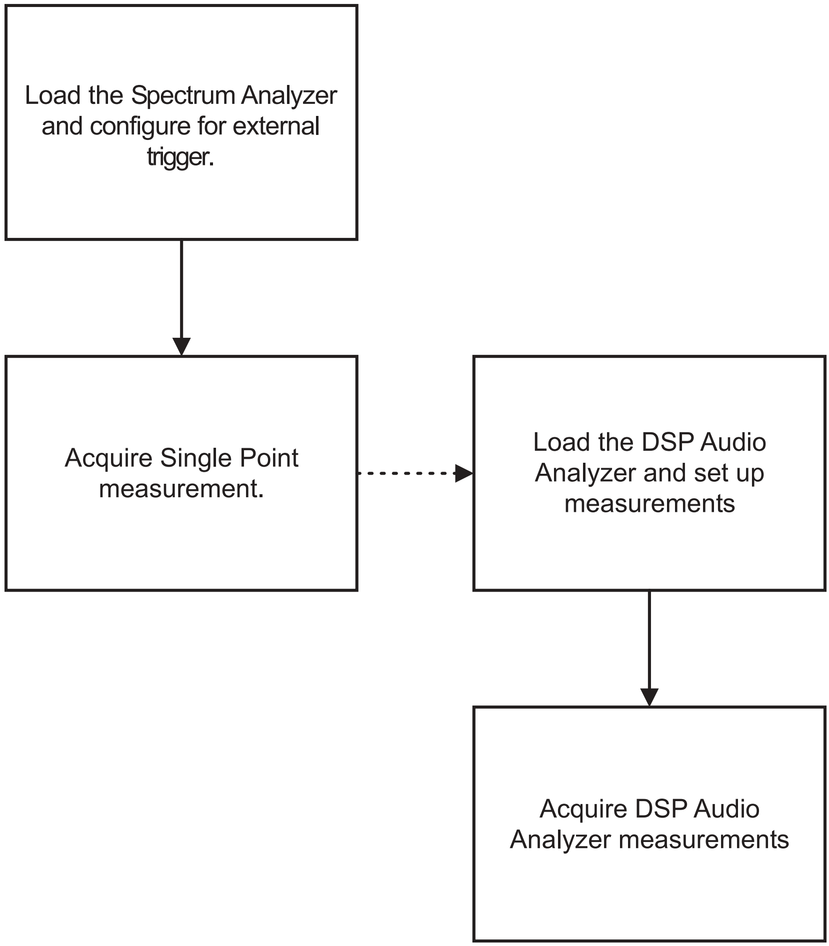


Figure 6. External trigger test sequence for DSP Audio Analyzer measurements.

Figure 7 shows the Sweep panel and FFT panel setup required to achieve a very fast acquisition with external triggering. In order to reduce acquisition time to a minimum, acquire a single point/single channel waveform with the sweep panel and do not display the graph. Configure the Spectrum Analyzer panel as follows: set **Acquire** to an acquisition length of **800** points, **Wave Display** to **Display Samples**, **Trigger Delay** to the value appropriate to avoid the mute interval in the test signal, **Trigger Source** to **External**, and **Trigger Slope** to **Pos**. The trigger slope setting depends on the nature of the trigger signal generated by the RF generator.

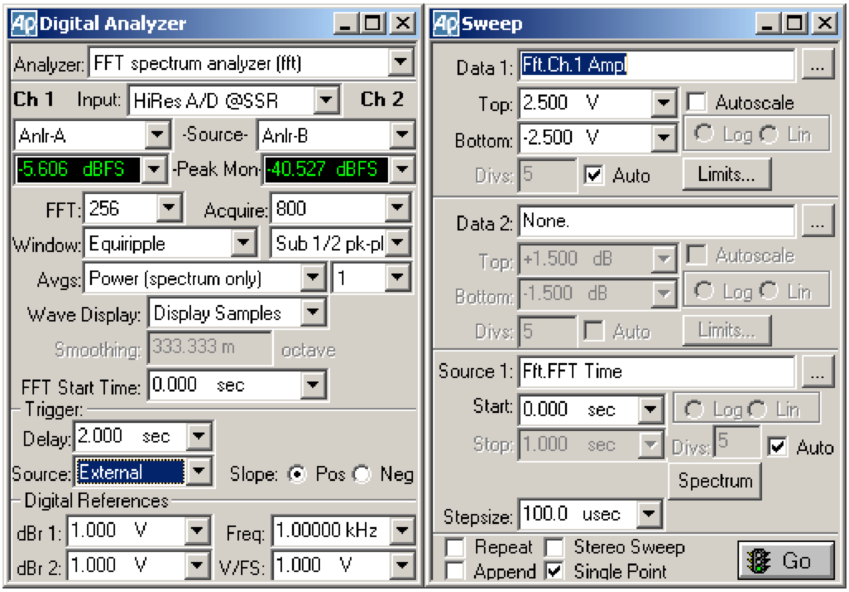


Figure 7. 2700 series Sweep and Spectrum Analyzer panel setup for a fast external trigger.

External Trigger Test Macro Example

An AP Basic macro that implements DSP Audio Analyzer measurements with external triggering is provided on the CD-ROM accompanying this application note. Open the macro “DAB Rcvr Audio Test - External Trigger Sample.apb” to review the source code, and run it. Edit the macro to change the trigger delay or other instrument settings for your application.

Audio Mute Detection

Audio mute detection is an alternative to an external trigger signal from the RF generator. This technique is applicable if external triggering is impractical but is more prone to false triggering on the mute event. This method uses a fast audio level meter within the analyzer.

Audio mute detection is feasible if the mute interval is long compared to the reading rate of the analyzer. The 2700 series and ATS-2 are capable of more than 128 readings per second with the DSP Analyzer level meters if level settling is disabled and auto ranging is disabled. Mute intervals under 250 ms are detectable with this method.

The expected audio level must be measured continuously in a software loop and compared against a lower and upper threshold level. The algorithm involves acquiring a level measurement in a continuous loop until it is below the lower threshold limit, and then acquiring successive level measurements until a measurement is above the upper threshold limit. The loops should include a time limit to prevent an endlessly repeating loop if the signal level does not transition through the two threshold levels. The signal level has returned to an acceptable level when the measurement has returned above the upper threshold limit.

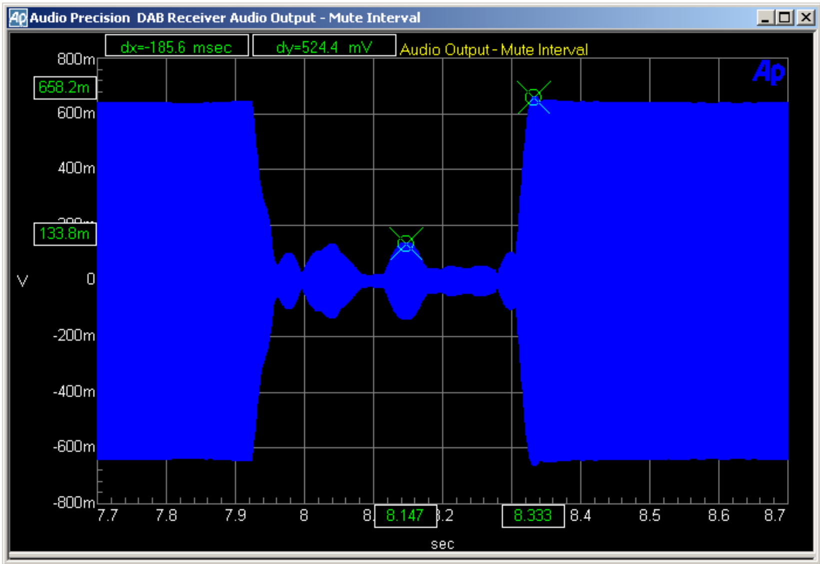


Figure 8. 2700 series FFT waveform. The cursors show the mute detection level threshold extremes for this DAB receiver program channel.

Figure 8 illustrates the two peak levels on the audio envelope that would be threshold extremes. Cursor 1 is the “maximum” low extreme of 0.1338 Vp (0.095 Vrms). Cursor 2 is the high extreme of 0.6582 Vp (0.465 Vrms). The difference between the two extremes is 0.5244 Vp (0.371 Vrms).

The lower detection threshold should be above the maximum low extreme by about 20% of the difference between the two extremes. This would be 0.239 Vp (0.169 Vrms) in this case (0.1338 Vp + (0.2 x 0.5244 Vp)).

The upper detection threshold should be below the upper extreme by about 20% of the difference between the two extremes. This would be 0.553 Vp (0.391 Vrms) in this case (0.6582 – (0.2 x 0.5244)).

This technique is sensitive to gain errors in the receiver; threshold values should be adjusted to deal with the worst case amplitude extremes expected in the receiver outputs due to component tolerances. Stereo DSP Audio Analyzer measurements of level and THD+N may be acquired in less than 2 seconds after the mute interval.

The signal of interest must first be measured to determine the nature of the mute interval. Do this by using the FFT analyzer to acquire a waveform with a large acquisition length. The sweep data in Figure 8 was acquired with this method. The Audio Precision 2700 series has the ability to acquire up to 4194303 points of stereo waveform data, 64 seconds at a sample rate of 65536. Use the 2700 series test “DAB Receiver Audio Output Waveform.at27” (provided on the CD-ROM with this application note) to acquire a 32-second waveform envelope.

Audio Mute Detection Test Macro Example

An AP Basic test macro that implements DSP Audio Analyzer measurement with mute detection is provided on the CD-ROM accompanying this application note. Load the macro “DAB Rcvr Audio Test - DSP Audio Analyzer Mute Detection Sample.apb” to review the source code, and run it. The threshold values used in the macro are described above. Edit the macro to change the thresholds or other instrument settings for your application.

Audio Tests with the DSP Audio Analyzer

The most common audio test signal is a sine wave at a fixed frequency and amplitude. Tests with the DSP Audio Analyzer typically include level at frequency, stereo level balance, crosstalk, distortion, noise, and dynamic range for A/D converter outputs. Although the Analog Analyzer can make these measurements, the DSP Audio Analyzer is a better choice because it can be used for both analog and digital inputs and measures both left and right channels simultaneously.

The receiver’s analog output level may be measured relative to its full scale output if the D/A convert full scale output level is known. This is accomplished by setting the analyzer’s dBr reference level to the full scale output level and using dBr units for analog level measurements.

Spectrum Analysis with the FFT Analyzer

Triggered spectrum measurements are possible by using the FFT Analyzer with an external trigger signal (the same method discussed above). This method is demonstrated with the test file “DAB Receiver Audio Output FFT Spectrum.at27 “ provided on the CD-ROM.

Figure 9 shows the FFT panel setup for external triggering with a 2-second acquisition delay after the external trigger event. The FFT Start Time should be set to the same value as the Trigger Delay.

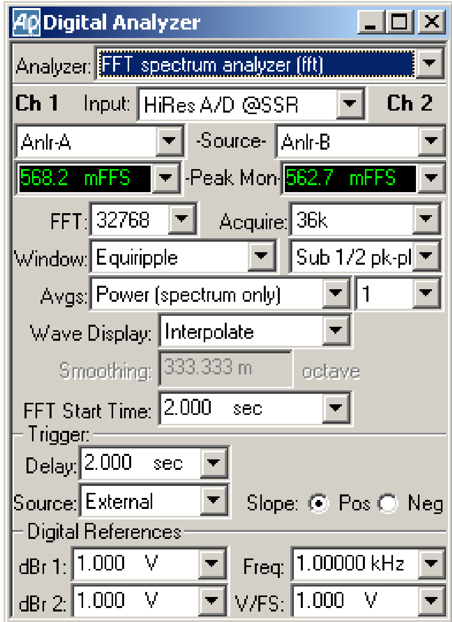


Figure 9. 2700 series digital analyzer panel. The FFT Spectrum Analyzer is configured for an

The FFT spectrum graph display that is shown in Figure 10 is a successful external triggered acquisition.

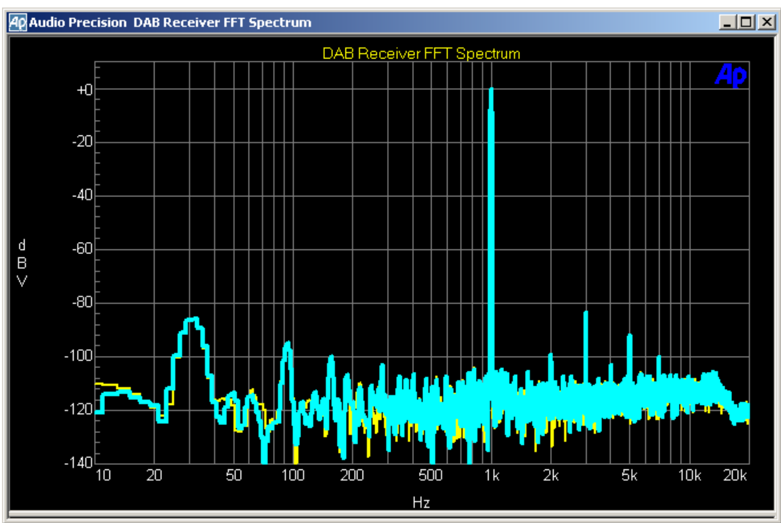


Figure 10. 2700 series Analyzer FFT spectrum of a DAB receiver program channel after an external trigger from an RF generator.

The audio mute detection method may also be used to synchronize signal acquisition and measurement with the FFT spectrum analyzer. Use the DSP Audio Analyzer to detect the mute event and then immediately switch to the FFT analyzer for a free-run triggered sweep. The mute detection method for FFT acquisition is demonstrated with the test macro file "DAB Rcvr Audio Test - DSP Audio Analyzer Mute Detection FFT Spectrum Sample.apb" provided on the CD-ROM.

Faster Tests with the Multitone Analyzer

Synchronizing audio testing with the RF generator mute interval to produce reliable repeatable measurements is only one of the potential problems encountered in testing DAB receivers. Testing time is also of concern, particularly to a test engineer developing a production test QA procedure.

A comprehensive audio test of a DAB receiver should test the full audio pass band. However, a large number of single tone sine wave test program channels are required because of the limitations of the RF generator. As mentioned previously this could take a substantial length of time to accomplish using 20 or more program channels with up to 10 seconds per channel.

A multitone test signal is a good alternative to single-tone sine waves for reduced testing time because only three receiver program channels are required. Use the Multitone Analyzer feature of the 2700 series and ATS-2 for level, frequency response, stereo level balance, and crosstalk with only three acquisitions.

Triggered multitone measurements are implemented with the same method used for triggered FFT measurements. The Multitone Analyzer panel must be set to use the external trigger signal. A delay setting provides the opportunity to set the trigger delay as well. As with the FFT Spectrum Analyzer, the mute detect method can also be used provided the Multitone Analyzer Triggering control is set to "Off" to permit free-run triggering.

Develop Multitone Signals with the Multitone Creation Utility

A multitone signal is, by definition, a specific set of sine wave amplitudes and frequencies mixed into a composite signal with a specific stereo balance and peak amplitude. How do you create such a signal? Use the Audio Precision Multitone Creation Utility include with the 2700 series or ATS-2. Refer to the tutorials provided with the software for more information.

In order to measure a multitone signal from a DAB receiver with the Audio Precision audio analyzers, an identical digital waveform must be loaded into the digital or analog arbitrary generator. This should be the same digital waveform that was used to create the receiver test signal. The multitone audio analyzer DSP program uses this generator waveform during measurement processing. The Multitone Creation utility will create the necessary files for use with the 2700 series or ATS-2 and will also load the instrument's analog or digital generator to generate the signal in real time for recording a test signal for the receiver. The utility will also digitally create a PC waveform file that may be used with hard disk audio editing systems.

The CD-ROM for this application note contains four sets of multitone waveforms created with the utility for sample rates of 32000, 44100, 48000, and 65536. Each waveform contains frequencies from 20 Hz to 20 kHz (16 kHz for the 32K sample rate) in one-third octave steps. Each waveform tone is approximately –25 dBFS. The waveforms have a crest factor (ratio of peak level to rms level) of about 3.8 with a peak headroom of –1 dB. Adjust the output level of the generator to lower the level of the waveform when recording digital test signals for the receiver. The set of waveform files includes the waveform to be loaded into the arbitrary generator in the instrument, sweep files, and data files that provide additional information about the technical attributes of the waveforms. Generate crosstalk signals by muting one of the generator output channels when you record test signals for the receiver.

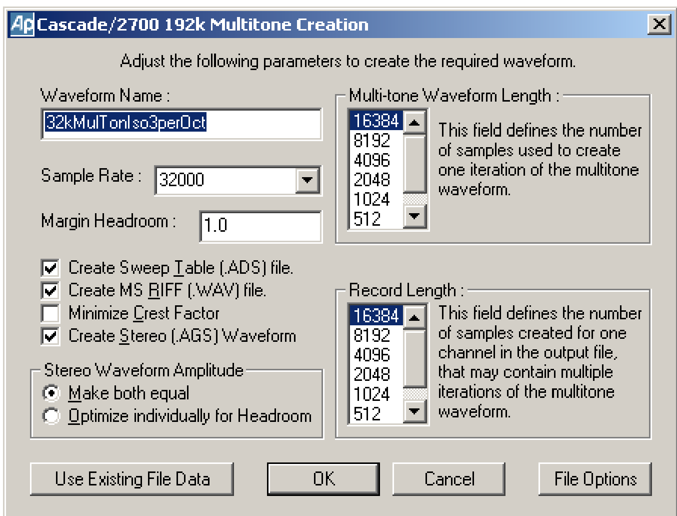


Figure 11. 2700 series Multitone Creation utility menu.

Synchronous Multitone Measurements Improve Accuracy

Synchronous multitone measurements are desirable in order to improve amplitude measurement accuracy and increase frequency resolution and dynamic range in the measurement results. The term synchronous applies to the sample rate and the frequencies used in the multitone signal. A synchronous frequency is an integer multiple of the instrument's arbitrary generator sample rate divided by the arbitrary generator record length.

The Multitone Creation utility automatically converts the frequencies you specify into synchronous frequencies. For example, if the Multitone Creation utility is used to create a multitone with a sample rate of 44.1 kHz and a record length of 8192, a synchronous frequency would be one that is an integer multiple of 44100 / 8192, or 5.3833 Hz. This is the basic frequency resolution for this particular combination of sample rate and generator record length. If you want one of the multitone frequencies to be close to 1000 Hz, then divide 1000 Hz by the basic resolution, round to an integer, and multiply by the basic resolution. For example, if the basic resolution is 5.3833 Hz, a synchronous frequency near 1000 Hz would be 1001.2938 (Round(1000/5.3833) * 5.3833).

Synchronous processing is not possible with digital outputs from a DAB receiver if the receiver's output sample rate is different from the rate used to create the multitone signal. This is possible because the generator signal may have been sample-rate converted to a different rate during the production of broadcast program material. For example, SDARS systems use a 32 kHz sample rate and must perform rate conversion on program material recorded at 44.1 kHz. Take care to produce multitone signals with the same sample rate as the receiver digital output if you want the benefits of synchronous multitone analysis for digital audio output testing. Common digital output sample rates are 24 kHz, 32 kHz, 44.1 kHz, and 48 kHz. Non-synchronous multitone measurements must be used if there is a sample rate conflict. Select Windowed Processing on the Multitone Analyzer panel in this case, but expect slightly less accurate amplitude measurements.

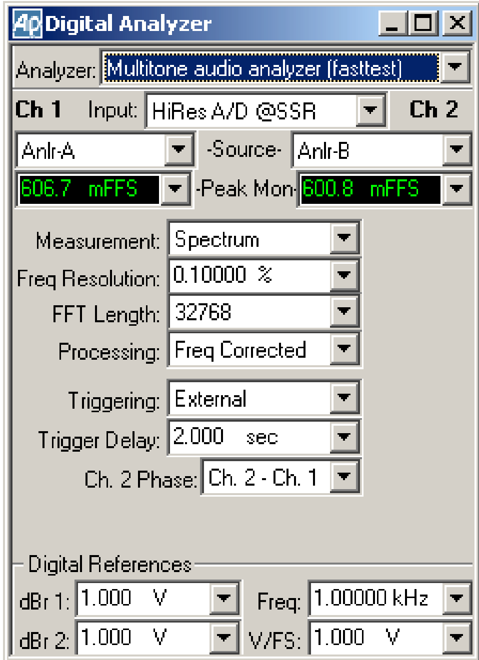


Figure 12. This 2700 series Multitone Analyzer panel is configured for a 2-second delayed

The Multitone Audio Analyzer Freq Resolution and Processing settings are of particular interest with regard to synchronous multitone measurements. Figure 12 shows the Multitone audio analyzer panel with a Freq Resolution setting of 0.1% and Processing setting of Freq Corrected. The Processing settings Freq Corrected and Synchronous both permit measurements of multitone signals without the use of an FFT window function. This eliminates measurement errors that would occur if an FFT windowing function were used. The Freq Corrected setting performs sample rate correction on the acquired signal within the percent setting of the Freq Resolution control to adjust for small frequency shifts in the receiver audio signal due to receiver sample rate clock errors or other transmission clock errors. The Freq Resolution setting may be adjusted up to 7%.

The Freq Resolution setting affects another post-acquisition process, causing a summation or exclusion of spectrum measurement bins above and below a frequency bin selected during a multitone sweep. This feature was provided for a tape recording system with significant wow and flutter but is not desirable for DAB receiver testing. For spectrum amplitude measurements this may cause an error in amplitude if the Freq Resolution % setting is particularly high because adjacent frequency amplitude bins will be summed resulting in a higher amplitude than that of the selected bin. Avoid this by setting Freq Resolution to 0% after the initial acquisition with the F9 function key (or the AP basic AP.Sweep.Start command) and then reprocess the data with the F6 function key (or the AP basic AP.Sweep.Reprocess command).

Test Stereo Level and Frequency Response with a Multitone Signal

Use a single stereo multitone signal containing all of the frequencies of interest to test level and frequency response with one triggered acquisition. Figure 13 shows both the level curves and the audio spectrum of a stereo 7 tone multitone signal acquired from an SDARS receiver.

The upper traces (cyan and blue) are level measurements at the synchronous multitone frequencies only. The level measurements are processed using a sweep file that specifies the exact frequencies of the multitone signal. A limit file may be produced using the same frequencies with appropriate upper and lower acceptance limits. The spectrum data (yellow and green traces) is not needed to perform the level measurement but is shown for clarity.

The only difference between the level measurements and the spectrum measurements is the number of frequencies used to extract data from the instrument after the initial acquisition. For the level measurements, a sweep file of only 6 frequencies was used to process the original acquisition data (the frequencies at 2.5 kHz and 5 kHz were omitted). For the spectrum display, a larger number of frequencies were specified on the Sweep panel with starting and ending frequencies set at 10 Hz and 22.5 kHz. Automatic data peak-picking algorithms assure that the highest signals found between any two sweep frequencies are used for the FFT curve data in order to show all signals within the spectrum.

Frequency response (not shown) would be measured relative to the amplitude of a reference frequency, usually 1 kHz. This is easily measured by post processing the level curve data with the Compute Normalize function.

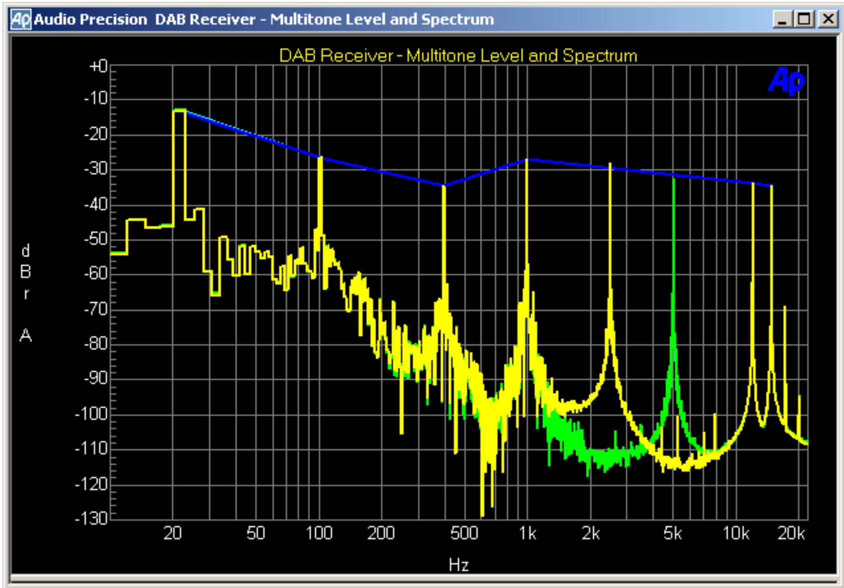
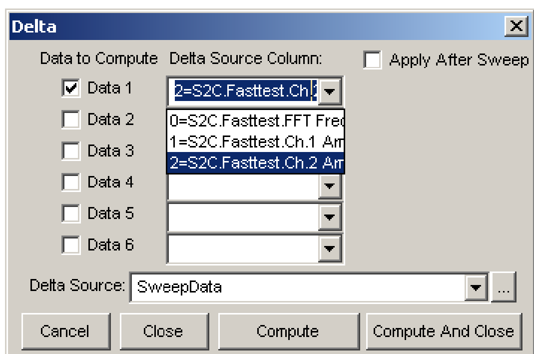


Figure 13. 2700 series Multitone Analyzer Spectrum and Level graph. The level is plotted using a sweep file containing multitone frequencies common to both channels.

Test Multitone Stereo Level Balance with Compute Delta Function

Multitone stereo level balance is the difference between the left and right channel levels at each frequency. Stereo level balance is easily tested with the Compute Delta function of the 2700 series and ATS-2.



The Compute Delta function may be used to produce a graph trace that is the difference between the two channels as a function of frequency. Figure 14 shows the results of using the Compute Delta function to process the two level curves shown in Figure 13. Channel B was compared with channel A. The graph shows that channel B is slightly higher in amplitude, particularly at the extremes of frequency, yet overall the level balance is within 0.1 dB from 20 Hz to 14.8 kHz.

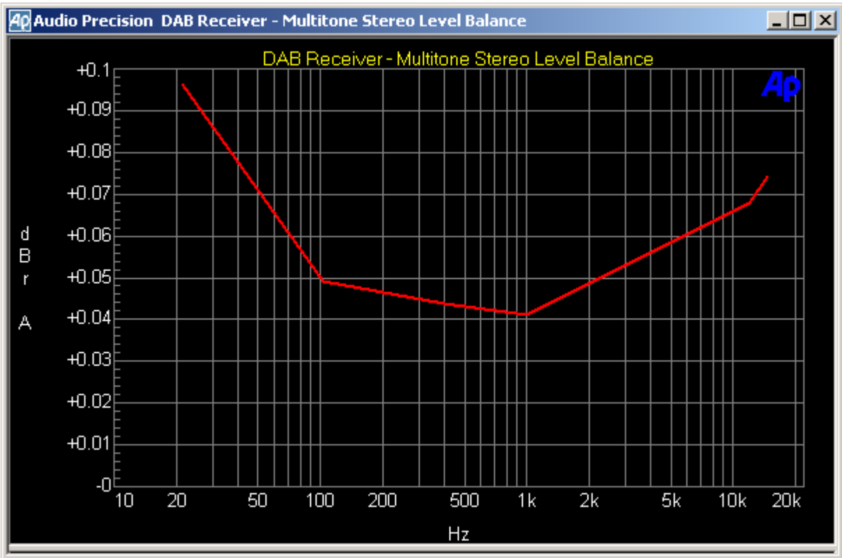


Figure 14. 2700 series Multitone Analyzer stereo level balance measurements produced with the Compute Delta function. The red trace is the difference in dB between channel A and B as a function of frequency.

Testing Stereo Crosstalk with a Multitone Signal

Stereo crosstalk measurements of a DAB receiver are usually performed by driving only one channel with a sine wave at a specific frequency and measuring the amplitude of the other stereo channel at the same frequency with a tuned bandpass filter, then reversing the process for the opposite channel. With a single tone signal this would require testing with two DAB receiver program channels per frequency of interest (for example 7.5 kHz channels 14 & 15 in the test channel plan above). This would require up to 10 seconds per test, 20 seconds per frequency for crosstalk A to B and B to A (assuming external triggering is required due to the receiver audio mute problem). Given seven frequencies of interest with the test channel plan above, the total test time could be as high as 140 seconds.

Reduce Crosstalk Test Time with a Full Bandwidth Multitone Test Signal

You may reduce crosstalk test time significantly using two multitone signals that contain all of the frequencies of interest. One DAB receiver program channel should use a multitone signal with all frequencies in the left channel and no signal in the right; another program channel should use a multitone signal with all frequencies in the right channel and none in the left. Only two multitone measurement acquisitions are necessary for crosstalk left-to-right and right-to-left, resulting in a maximum triggered test time of 20 seconds.

Figure 15 shows an SDARS receiver multitone crosstalk measurement of a program channel with a 7 tone multitone signal in the left stereo channel but no signal in the right. The difference between the two crosstalk curves (left = red, right = blue) has been computed and entered into the graph comment field for clarity. Note that crosstalk increases at the higher frequency 12 kHz and 14.8 kHz tones.

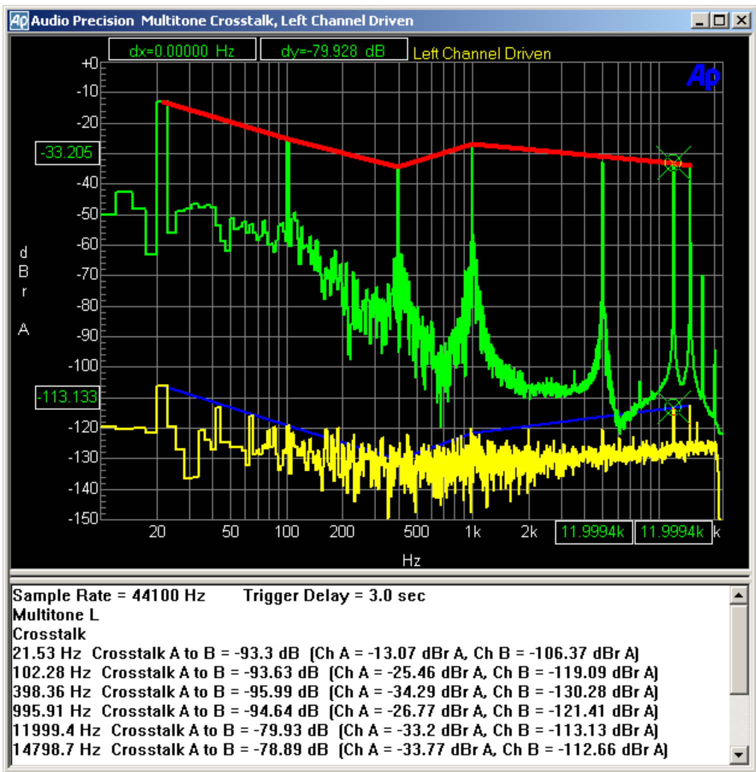


Figure 15. 2700 series Multitone Analyzer crosstalk and spectrum measurements of an SDARS receiver with Left channel driven, Right channel muted, cursors at the 12 kHz tone.

Conclusion

The two measurement synchronizing techniques for Audio Precision audio analyzers discussed in this application note solve the problem of unreliable DAB receiver audio measurements caused by periodic audio muting when the receiver is stimulated with an RF generator. The external trigger method was used to trigger and delay audio analyzer measurements with a trigger signal from the RF generator in order to avoid the mute. The audio mute detection method provides reliable measurements when an external trigger from an RF generator is unavailable and receiver test signal attributes are well known.

DAB receiver test time is lengthy if numerous single-tone test channels are used. You may use multitone signal measurement techniques to reduce this test time significantly, however, for level, stereo balance, and crosstalk tests. You should use these in conjunction with single-tone signals for distortion and dynamic range tests.

Application Software Puts It All Together

Application software is provided on the accompanying CD-ROM to get you started with a 2700 series or ATS-2 instrument.

DAB Receiver Audio Test Menu

Use the DAB Receiver Audio Test Menu macro to test a receiver channel. The menu selections allow you to setup the test for type of signal, analog or digital audio format, measurements, and additional measurement attributes. Measurement data will be saved in a text log file and test files for later use. This macro is available on the accompanying CD-ROM in two forms, one for each type of instrument (2700 series or ATS-2).

To use these AP Basic macro, copy the software directory “DAB Receiver Audio Test” from the CD-ROM into a new folder on your hard disk with Windows Explorer. Power up and connect your 2700 series or ATS-2 instrument to the APIB interface on your computer. Run the control software and use the File menu to open the macro file “DAB Rcvr Audio Test Menu”.

Run the macro from the Macro menu. When the panel shown in Figure 16 appears, select the Help button to browse a Windows help file that describes how to use the DAB Receiver Audio Test. The help file may also be accessed directly from the CD-ROM with Windows Explorer.

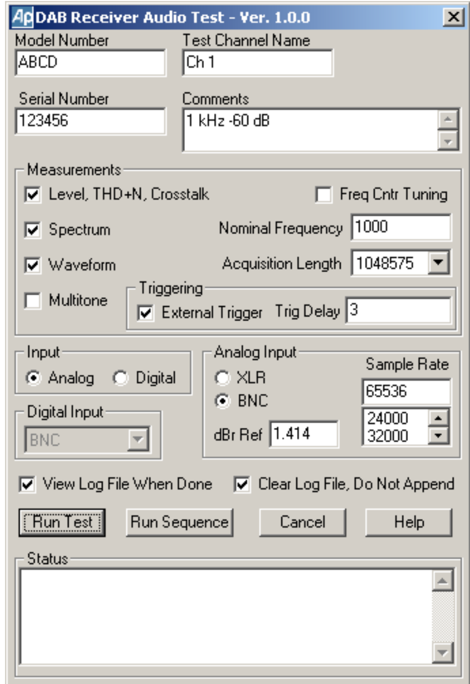


Figure 16. The DAB Receiver Audio Test control panel for Audio Precision Series 2700 analyzers. Setup the attributes of your tests and select the Run

Enter data about your receiver and the receiver test signal program channel in the text boxes. Then select the measurements you want and set related parameters. Select the appropriate analyzer input settings and log file settings.

Run your test with the Run Test button. The test results log file, waveforms, and test files will be saved in the directory containing the test macro. The log file contains the names of the test and waveform files saved during the test. See Help for more information.

Use the Run Sequence button to specify a test sequence file and run a sequence of tests specified in the file. The test sequence file specifies a list of DAB receiver program channel names. The list consists of text lines with comma-separated fields. Each field on a line contains information relevant to a receiver program channel, such as signal type, frequency, analog reference level, and test limits.

The default test sequence file is an example only and must be edited to contain field data relevant to the receiver program channels encoded for your RF generator. The default sample CSV file is provided to illustrate how data should be entered into each line of the file. The file may be edited with any text editor or with Microsoft Excel. Develop a unique test sequence file for each model or version of a receiver containing test parameters unique to the receiver.

Develop Your Test Macros with the Audio Test Library

The audio test methods described in this application note have been implemented in two libraries of AP Basic functions on the CD-ROM in the macro files “DAB Rcvr Audio Test Library.apb” and “DAB Rcvr Audio Test Library.atsb” (one for the 2700 series one for ATS-2). These files are used by the DAB Receiver Audio Test Menu macros for the 2700 series and ATS-2 instruments.

The macro files “DAB Rcvr Audio Test Example.apb” and “DAB Rcvr Audio Test Example.atsb” illustrate how to use the libraries to develop your own DAB receiver testing macros.

Files on the CD-ROM

The files provided on the CD-ROM with this application note are divided into two groups depending on the type of instrument/control software you are using. This table identifies files associated with an instrument type and its control software.

File Type	Filename Extension	Instrument / Control Software
AP Basic macro files	*.apb	2700 series / AP2700
Test file	*.at27	2700 series / AP2700
Sweep file	*.ads	2700 series / AP2700
AP Basic macro files	*.atsb	ATS-2 / ATS
Test file	*.ats2	ATS-2 / ATS
Sweep file	*.atss	ATS-2 / ATS

CD-ROM File Path	Description
\DAB Rcvr Audio Test\DAB Rcvr Audio Test Menu.apb \DAB Rcvr Audio Test\DAB Rcvr Audio Test Menu.atsb	This is an AP Basic test macro with a graphical user interface to set measurement options, run tests on receiver program channels, and run sequences of receiver program channel tests with test limits. See Figure 16 (2700 version). Load this macro and Run it from the Macro menu.
\DAB Rcvr Audio Test\AUDIO PRECISION DAB RCVR AUDIO TEST AP2700.HLP \DAB Rcvr Audio Test\AUDIO PRECISION DAB RCVR AUDIO TEST ATS.HLP	This is the Windows help file for “DAB Rcvr Audio Test Menu.apb” and “DAB Rcvr Audio Test Menu.atsb” macros.
\DAB Rcvr Audio Test\DAB Rcvr Audio Test Example.apb \DAB Rcvr Audio Test\DAB Rcvr Audio Test Example.atsb	This is an AP basic example program to illustrate how to use the function library for a single set of tests on one receiver program channel. Load this macro and Run it from the Macro menu.

\DAB Rcvr Audio Test\DAB Rcvr Audio Test Library.apb \DAB Rcvr Audio Test\DAB Rcvr Audio Test Library.atsb	This is an AP basic library of functions and structures used to test DAB receivers for Level, THD+N Ratio, and Crosstalk. It acquires stereo FFT spectra, waveform envelopes, and multitone measurements. This library uses the external trigger method of measurement synchronization with the DAB receiver RF generator. Comprehensive data logging features are supported. Functions in this file must be called from other macros.
\DAB Rcvr Audio Test\DAB Rcvr Audio Test 1.at27 \DAB Rcvr Audio Test\DAB Rcvr Audio Test 1.ats2	2700 series and ATS-2 test files for fast single-sample waveform sweep with the FFT Analyzer; use for fast external trigger prior to DSP Analyzer measurements. Used by “DAB Rcvr Audio Test Library.apb” and “DAB Rcvr Audio Test Library.atsb” test macros.
\DAB Rcvr Audio Test\DAB Rcvr Audio Test 2.at27 \DAB Rcvr Audio Test\DAB Rcvr Audio Test 2.ats2	2700 series and ATS-2 test files for FFT spectrum measurements. Used by “DAB Rcvr Audio Test Library.apb” and “DAB Rcvr Audio Test Library.atsb” test macros.
\DAB Rcvr Audio Test\DAB Rcvr Audio Test 3.at27 \DAB Rcvr Audio Test\DAB Rcvr Audio Test 3.ats2	2700 series and ATS-2 test files for Multitone measurements. Used by “DAB Rcvr Audio Test Library.apb” and “DAB Rcvr Audio Test Library.atsb” test macros.
\DAB Rcvr Audio Test\DAB Rcvr Audio Test 4.at27 \DAB Rcvr Audio Test\DAB Rcvr Audio Test 4.ats2	2700 series and ATS-2 test test files for Audio Analyzer measurements. Used by “DAB Rcvr Audio Test Library.apb” and “DAB Rcvr Audio Test Library.atsb” test macros.
\DAB Rcvr Audio Test\DAB Rcvr Audio Test Sequence.csv	CSV (comma separated values) file that may be edited by an ASCII file editor or MS Excel. This is a test sequence file used by “DAB Rcvr Audio Test Library.apb” and “DAB Rcvr Audio Test Library.atsb” test macros. Copy the file and edit the copies to create your own program channel test sequence files for different receivers and audio output formats. Do not change the field names in the first line.
\DAB Rcvr Audio Test\Waveforms	Multitone signal files for level, stereo balance and crosstalk, coded at four sample rates: 32k, 44.1k, 48k, and 65k. These arbitrary multitone waveform files (*.agm files) may be used to generate receiver audio test program channels. Sweep files and *.agm files must be used by the three test procedures (*.apb and *.atsb files) to measure corresponding multitone signals at the receiver outputs (see inline comments).
\External Trigger \DAB Rcvr Audio Test - External Trigger Sample.apb \External Trigger \DAB Rcvr Audio Test - External Trigger Sample.atsb	An AP basic test macro that implements Audio Analyzer measurements with external triggering.
\External Trigger \DAB Receiver Audio Output FFT Spectrum.at27 \External Trigger \DAB Receiver Audio Output FFT Spectrum.ats2	2700 series and ATS-2 test files that uses FFT Spectrum Analyzer with external trigger to measure a DAB receiver audio output spectrum.
\Mute Trigger\DAB Rcvr Audio Test - DSP Audio Analyzer Mute Detection Sample.apb \Mute Trigger\DAB Rcvr Audio Test - DSP Audio Analyzer Mute Detection Sample.atsb	An AP basic test macro that implements Audio Analyzer measurements with mute detection.

\Mute Trigger\DAB Rcvr Audio Test - DSP Audio Analyzer Mute Detection FFT Spectrum Sample.apb \Mute Trigger\DAB Rcvr Audio Test - DSP Audio Analyzer Mute Detection FFT Spectrum Sample.atsb	An AP basic test macro that implements FFT Spectrum Analyzer measurements with mute detection.
\Mute Trigger\DAB Receiver Audio Output Waveform.at27 \Mute Trigger\DAB Receiver Audio Output Waveform.ats2	2700 series and ATS-2 test files that uses the Spectrum Analyzer to acquire a long waveform envelope (32-second for 2700 series, 8-second for ATS-2). Use this to inspect the receiver output audio envelope for mutes.