TN133

HEARING AID TESTS
WITH APX500 AUDIO ANALYZERS

by Joe Begin

□ 2700 Series
☑ APx555
☑ APx585 Series
☑ APx525 Series
☑ APx515
☑ APx511

About This Technote

In this technote, we discuss industry standard tests used to characterize the electroacoustic performance of hearing aids (also referred to as hearing instruments) with Audio Precision APx500 Series audio analyzers. We also describe a set of APx project files developed to facilitate these measurements. These project files make use of the APx Hearing Aid plug-in, which adds some derived results that are specific to hearing aid test.

Hearing Aid Test Standards

This Technote describes hearing aid tests according to the following international and American national standards:

- IEC 60118-0:2015 Electroacoustics - Hearing aids - Part 0: Measurement of the performance characteristics of hearing aids.

- ANSI/ASA S3.22:2014 Specification of Hearing Aid Characteristics.

These two standards are fundamentally equivalent; the main difference between them is that ANSI/ASA S3.22 specifies tests within the frequency range of 200 Hz to 5 kHz, whereas this latest edition of IEC 60118-0 extends the frequency range to 8 kHz. In the measurements described below, the IEC 60118-0 frequency range of 200 Hz to 8 kHz is assumed to apply. For ANSI S3.22, the range could be reduced to an upper frequency of 5 kHz.

Measurement Challenges

Hearing aids are unique audio devices in the sense that both the input to the device and the output from it are acoustic signals. Hearing aids typically have one or more microphones, a power amplifier, a digital signal processor

and a loudspeaker integrated in a tiny battery-powered package. In addition, many hearings aids have a Telecoil mode, in which the input to the device is a magnetic field instead of a microphone. These factors complicate the testing of hearing aids.

Basic Equipment needs

To measure the sound level output from the hearing aid, an acoustic coupler is required, either a 2 cc coupler (Figure 1) per IEC 60118-5 (or ANSI S3.7), or optionally, an occluded ear simulator (Figure 2) per IEC 60118-4 can be used. A measurement microphone of the pressure response type is required with a 2 cc coupler, and occluded ear simulators typically have an integrated pressure response microphone. These measurement microphones also require a microphone power supply (or signal conditioner).

Figure 1. 2 cc couplers with microphone and dummy microphone.

A loudspeaker with suitable power amplifier is required to test the acoustic input of the hearing aid. Uniform sound pressure level versus frequency is required. Therefore, the loudspeaker should have a relatively flat frequency response from 200 Hz to 8 kHz and the ability to generate sound levels of at least 90 dBSPL at the hearing aid test point, with low distortion.

Figure 2. An occluded ear simulator (per IEC 60318-4) with attached microphone preamplifier.

To test the Telecoil mode of the hearing aid an induction loop is required, as well as a suitable power amplifier to drive the loop.

An acoustic test chamber is also required, to isolate the hearing aid from ambient acoustic noise and to reduce un-wanted acoustic reflections. One example of a hearing aid test box that is widely used is the Interacoustics TBS25 (Figure 3). It provides up to 40 dB of acoustic noise isolation, is anechoic at frequencies above 500 Hz and includes a loudspeaker with flat frequency response (within a few dB) from 50 Hz to 8 kHz. This chamber also has a built-in induction loop for conducting Telecoil measurements and patch panels for conveniently routing connections from out-side the chamber to microphones, the hearing aid's power supply (battery simulator) and programming interface.

An audio analyzer is also required for hearing aid tests. For convenience, the analyzer should have built-in features such as microphone calibration, regulation and output equalization to enable calibrating the acoustic output and creating a uniform sound field. The audio analyzer should also have a wide selection of built-in audio measurements such as frequency response, level sweeps, harmonic and intermodulation distortion, etc., and the ability to generate a variety of audio signals, including sine waves, noise and speech.

Measurements

The measurements covered by this Technote and the ac-companying APx project files are listed in Table 1 with the corresponding numbered paragraphs in IEC 60118-0 and ANSI/ASA S3.22. These measurements are described be-low, in the order in which they appear in the APx project files.

Figure 3. An Interacoustics TBS25 Test Chamber.

Table 1. Measurements

Measurement	IEC 60118-0	ANSI/ASA S3.22
OSPL90 frequency response	7.2	6.2
Full-on Gain frequency response	7.3	6.4
Basic frequency response	7.4	6.8
Total Harmonic Distortion	7.5	6.11
Equivalent Input Noise	7.6	6.12
Steady State Input Output	9.8.2	C.9.1
Attack and Release	9.8.3	C.9.3
Intermodulation Distortion	9.3	C.14
Battery Current	7.7	6.13
SPLIV Curve	-	6.14.4.2
Equivalent Test Loop Sensitivity	7.8.2	6.14.4.4
HFA MASL	7.8.3	-

OSPL90 Frequency Response

OSPL90 stands for Output Sound Pressure Level at 90 dBSPL input. The sound pressure level (SPL) produced by the hearing aid when set to full-on gain (FOG) is measured in a coupler for an input SPL of 90 dBSPL, swept from 200 Hz to 8 kHz. Primary results include the OSPL90 fre-quency response curve and the HFA OSPL90.

HFA is an acronym for High Frequency Average. It is defined as the average of the level (in dBSPL) or gain (in dB) frequency response of the hearing aid at three specific frequencies: 1.0, 1.6 and 2.5 kHz. Hence HFA OSPL90 is the high frequency average of the OSPL90 frequency response curve at these frequencies.

Full-on Gain (FOG) Frequency Response

The Full-on Gain frequency response is the SPL produced by the hearing aid and measured in a coupler for an input SPL of 50 dBSPL, swept from 200 Hz to 8 kHz, with the hearing aid set to FOG. The acoustic gain is determined as the output SPL relative to the 50 dB input SPL. Primary results include the maximum full-on gain and the HFA-FOG.

Basic Frequency Response Curve

For this measurement, the hearing aid is set to a special gain setting known as the Reference Test Setting (**RTS**). RTS is defined as the gain setting for an input SPL of 60 dB at which the HFA-gain, as measured above, is within 1.5 dB of HFA-OSPL90 − 77dB. If the HFA-gain at FOG for an input of 60 dBSPL is less than HFA-OSPL90 − 77dB, then RTS is at the full-on gain setting.

Modern digital hearing aids can typically be set to RTS by a command sent over a hearing aid programming interface. For older analog hearing aids, RTS must be determined iteratively by repeatedly adjusting the hearing aid's volume control and measuring the HFA-Gain until RTS is achieved.

With the hearing aid at RTS, the Basic Frequency Response is measured as the SPL produced by the hearing aid in the coupler for an input SPL of 60 dB, swept from 200 Hz to 8 kHz. Primary results include the **Frequency Range** and the **Reference Test Gain**.

The **Frequency Range** is determined from the Basic Frequency response curve as follows (see Figure 4): First the HFA-Level is calculated from the curve. Next, frequencies f_1 and f_2 are found as the lowest and highest frequencies, respectively, at which the frequency response curve has the value of HFA-Level minus 20 dB. If f_1 or f_2 cannot be determined because they would fall outside the 200 Hz to 8 kHz frequency range, they can be shown as <200 Hz or >8 kHz.
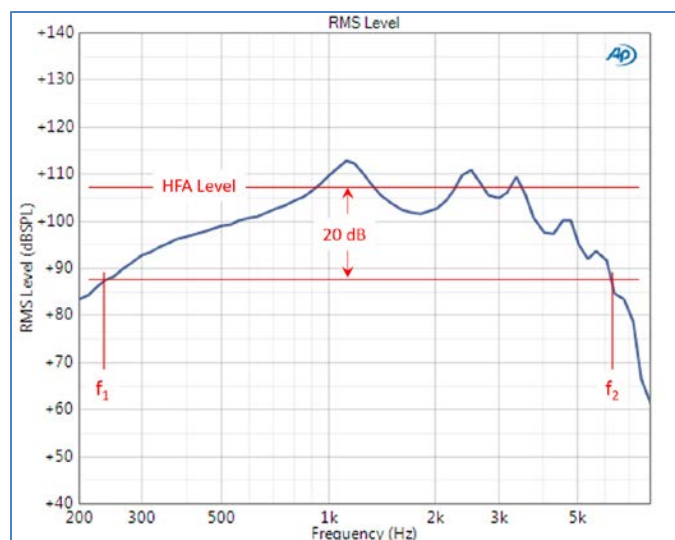

Figure 4. Basic frequency response curve (measured with 1/12-octave resolution) showing frequency range determination

The **Reference Test Gain** (RTG) is simply the HFA-Gain with the gain control at RTS and an input SPL of 60 dB. It can be derived from the Basic Frequency Response measurement data.

Total Harmonic Distortion (THD)

THD is measured in units of percent with the hearing aid at RTS at the frequencies and input SPLs shown in Table 2.

Table 2. Distortion test frequencies and levels

Frequency	Input SPL
500 Hz	70 dB
800 Hz	70 dB
1.6 kHz	65 dB
3.2 kHz	60 dB

Equivalent Input Noise (EIN)

Equivalent Input Noise is measured as follows: With the hearing aid at RTS, the HFA gain for an input SPL of 50 dB is measured (HFA-Gain50). Next, with the sound source switched off, the total output noise SPL is measured in the acoustic coupler as the band-limited (200 Hz to 8 kHz) noise level. EIN is calculated as total output noise SPL minus HFA-Gain50. Note: the ambient noise in the test space should be at least 10 dB lower than the output noise of the hearing aid.

Steady State Input Output

Steady state input output is a measurement used to characterize hearing aids that have automatic gain control (AGC). The SPL output by the hearing aid is measured as the input SPL of a sinusoidal signal is stepped from 50 dBSPL or lower to at least 90 dBSPL in steps of 5 dB or less. A sine frequency of 2 kHz is required, and other frequencies within the range of 200 Hz to 8 kHz are optional. The hearing aid output level vs. input level is plotted as shown in Figure 5.
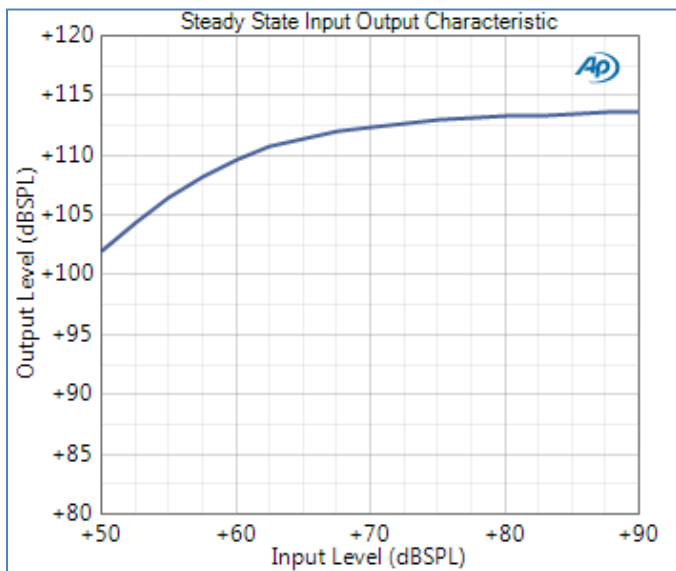

Figure 5. Steady state input output curve.

Attack and Release Time

This is another measurement used to characterize hearing aids with AGC. The hearing aid output SPL is measured in a coupler as a sinusoidal input signal is abruptly increased from 55 to 90 dBSPL (attack) and, after stabilizing, abruptly decreased from 90 to 55 dBSPL (release). The attack and release times are determined from the envelope of the measured signal. The attack time is the time that it takes for the measured level to stabilize within 3 dB of its steady state value following the attack step. The release time is the time that it takes for the measured level to stabilize within 4 dB of its steady state value following the release step. A measurement at 2 kHz is required; other sine frequencies in the 200 Hz to 8 kHz range are optional.

Intermodulation Distortion (IMD)

IMD is listed in section 9 of IEC 60118-0, which has the heading "Additional optional test procedures". The specified type of IMD is Difference Frequency Distortion (DFD).

Two sine signals at frequencies f_1 and f_2 are generated at equal levels such that the difference frequency ($f_2 - f_1$) is 125 Hz and the total level of the combined signals is 64 dBSPL. The frequencies are swept such that f_2 is varied from 350 Hz to 8 kHz, while maintaining a difference frequency of 125 Hz. The second and third order IMD components are measured (i.e., at frequencies $f_2 - f_1$ and $2f_1 - f_2$). An example of one hearing aid's DFD curve is shown in Figure 6.
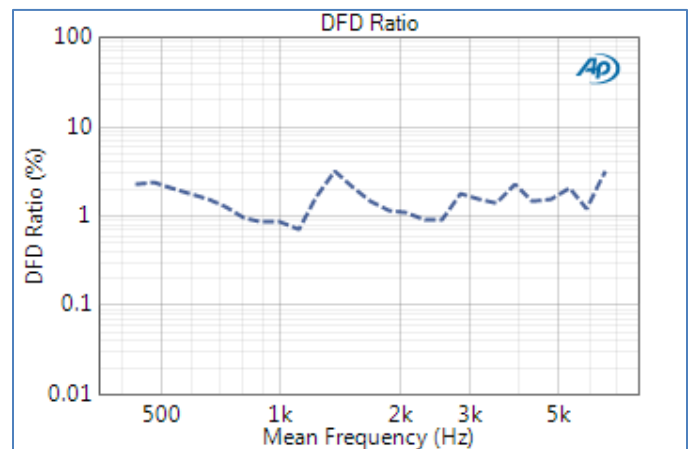

Figure 6. Difference Frequency Distortion vs Frequency.

Battery Current

Battery current is a measure of the dc current drawn by the hearing aid when it is set to RTS and is subject to a sinusoidal input of 65 dBSPL at 1.0 kHz.

SPLIV Curve

SPLIV is an acronym for Sound Pressure Level in a Vertical magnetic field. The hearing aid is subjected to a vertical magnetic field when it is oriented such that its vertical reference line (as specified by the manufacturer or determined by inspection) is aligned with the axis of the magnetic field created by the induction loop.

A loop current is established to create a sinusoidal magnetic field with a field strength of 31.6 mA/m. The loop current required to achieve the specified field strength depends on the loop; for the loop in the TBS25 test chamber (which is oriented left to right) a current of 7.9 mA is required.

With the hearing aid oriented as required, set to Telecoil mode, and its gain set at RTS, the loop sine frequency is swept from 200 Hz to 8 kHz, and the output SPL of the hearing aid is measured in the acoustic coupler.

Equivalent Test Loop Sensitivity

The Equivalent Test Loop Sensitivity (ETLS) may be determined from the SPLIV curve, measured as described above, by first determining the HFA-SPLIV in units of dBSPL. ETLS is calculated as HFA-SPLIV minus (RTG + 60 dB).

HFA MASL

HFA MASL is an acronym for maximum HFA Magneto-Acoustical Sensitivity Level. It is measured under the same conditions as the SPLIV curve, described above, except that the hearing aid is set to full-on gain and the magnetic field strength is set to 10 mA/m. First the HFA output SPL is measured and then HFA MASL is calculated as

$$HFA\ MASL = HFA\ OSPL - 20log_{10}\left(\frac{H}{1\frac{mA}{m}}\right)$$

where $HFA\ OSPL$ is the high frequency average OSPL and H is the magnetic field strength in mA/m.

Hearing Aid Tests with APx500

Acoustic Test Method

No loudspeaker has a perfectly flat frequency response, yet these standards require that the sound pressure level at the hearing aid test point be constant within specified limits, as the frequency is varied in the range from 200 Hz to 8 kHz. IEC 60118-0 specifies two alternative methods of achieving this required flat SPL response:

1. Constant entrance sound pressure method: A reference microphone is maintained in close proximity to the hearing aid test point. At each frequency, sound levels generated by the loudspeaker are adjusted to the required SPL, as measured with the reference microphone. A second test microphone is used in the acoustic coupler to measure the hearing aid output SPL. The name of this method is sometimes abbreviated to simply "pressure method".

2. Substitution method: The frequency response of the loudspeaker is first measured by the test microphone located in close proximity to the hearing aid test point. This response curve is stored and an inverted version of it is used to equalize the loudspeaker such that the sound field at the test point

has a flat frequency response. A "dummy microphone" (typically a block of aluminum) of the same size and shape as the measurement microphone is placed in the coupler for the initial frequency response measurement. For subsequent measurements, the positions of the test and dummy microphones are exchanged, to ensure that the sound field in the area of the test point is the same for both measurements.

The Substitution method has the advantage that only one measurement microphone (and one microphone power supply) is required. This is the preferred method for production test of hearing aids and is the use case around which the APx511 Hearing Instrument Analyzer was designed. Accordingly, the APx511 has one microphone input channel.

All APx audio analyzers (except the APx511) have at least two input channels and could therefore be used for the Pressure method.

Audio Analyzer Configuration

APx511 Hearing Instrument Analyzer

The APx511 Hearing Instrument Analyzer has been especially designed for hearing aid test, and is ideally suited to hearing aid tests using the Substitution acoustic test method. It has the following hearing aid test specific features:

- A class AB audio amplifier rated at 4 watts into 8 Ω.

- A current mode induction loop (Telecoil) amplifier rated at 110 mA into 4 Ω.

- Built-in constant current power (CCP) for a pre-polarized measurement microphone.

- A battery simulator power supply (including battery current measurement capability).

- A convenient cable for connecting the analyzer to an Interacoustics TBS 25 sound test chamber.

Other APx500 Audio Analyzers

Although not specifically designed for hearing aid test, the APx500 family has a number of other audio analyzers from which to choose, providing solutions for users requiring higher analog channel counts, higher analog performance, or access to a variety of digital I/O modules. Analog channel counts available include (output x input) 2x2, 2x4, 2x8,

8x8, and 8x16. Optional digital I/O modules available for all models (except APx515 and APx511) include Digital Serial (for I2S and TDM), Bluetooth, Pulse Density Modulation (PDM), HDMI and AES3-S/PDIF.

APx1701 Transducer Test Interface

When an audio analyzer other than the APx511 is used for hearing aid test, two power amplifiers are required—one to power the chamber's loudspeaker and the other to power the induction loop. A microphone power supply is also required.

The APx1701 Transducer Test Interface is an accessory for testing loudspeakers, headphones and microphones with APx audio analyzers. Although designed for testing electroacoustic transducers, its features can be used to advantage for hearing aid test as well. It has two class AB power amplifiers and two microphone input channels that provide constant current power for measurement microphones. It also comes standard with accredited calibration and is fully integrated with the APx500 control software.

Battery Simulator Power Supply/Circuit

For audio analyzers other than the APx511, a DC power supply that simulates the voltage and internal impedance of batteries should be used to power the hearing aid. An open circuit voltage of 1.3 Vdc is specified in IEC 60118-0 with battery internal impedances varying from 3.3 Ω to 8.2 Ω. A resistor in series with the power supply can be used to simulate the battery impedance, and battery current can be determined by measuring the voltage drop across this resistor. A balanced input should be used for this measurement, because the measured voltage drop is likely small (in the millivolt range).

> Note: If the APx audio analyzer is equipped with a PDM module and the required current does not exceed 15 mA, the module's Vdd power supply can be used to power the hearing aid.

> Note: An Audio Precision DCX-127 could also be used to power the hearing aid if the required current does not exceed 20 mA.

Induction Loop Resistor Circuit

For audio analyzers other than the APx511, an external power amplifier is required to drive the magnetic induction loop. An induction loop will probably have a lower impedance than a typical loudspeaker. For example, the loop in the TBS25 sound test chamber has an impedance magnitude of less than 2 Ω. A current amplifier, like the one in the APx511, can be used to drive the induction loop. Alternatively, an ordinary voltage mode audio amplifier can be

used with a load resistor connected in series with the loop. IEC 60118-0 recommends using a series resistor with an impedance at least 100 times greater than the input impedance of the loop (or magnetic field source) in the frequency range from 200 Hz to 8 kHz. The voltage drop across this series resistor can be used to measure and regulate the audio analyzer's generator output level to achieve the required magnetic field strength. In the APx500 software a generator reference known as dBRG is available for this purpose. The Auto Set Generator level feature of the software can be used to set the dBRG reference such that 0 dBrG corresponds to the generator voltage needed to achieve a magnetic field strength of 31.6 mA/m. Once set, the generator level can conveniently be set in units of dBrG.

APx Project Files

This Technote is accompanied by three APx500 project files. Each is meant to work with a different hardware configuration, as follows:

1. An APx511 Hearing Instrument Analyzer
2. Any APx audio analyzer (other than APx511) *with* an APx1701 Transducer Test Interface
3. Any APx audio analyzer *without* an APx1701 Transducer Test Interface

Different equipment accessories are required depending on which analyzer configuration is chosen. Equipment requirements are summarized in Table 3.

Connections

Connections from the audio analyzer to the hearing aid test chamber are different, depending on which audio analyzer is used and whether the APx1701 Transducer Test Interface is used.

> Note: The project files accompanying this Technote have been created with the assumption that all necessary connections to the audio analyzer for acoustic, telecoil and battery current measurements will be made before running either of the two sequences included. If this is not the case, the sequences will not run as expected.

APx511 Hearing Instrument Analyzer

Configuring the APx511 for hearing aid tests is straightforward. The test microphone should be connected to the microphone input BNC connector on the back of the analyzer, and the breakout cable provided has connections for the chamber's loudspeaker, induction loop jack and battery simulator jack.

Table 3. Equipment requirements by analyzer configuration (✓ denotes item is required)

Equipment Item	APx511 Hearing Instrument Analyzer	APx Audio Analyzer **with APx1701**	APx Audio Analyzer **without APx1701**
Test Chamber	✓	✓	✓
Acoustic Coupler	✓	✓	✓
Measurement Microphone	✓	✓	✓
Dummy Microphone	✓	✓	✓
DC Power Supply with Resistor Circuit		✓	✓
Induction Loop Resistor Circuit		✓	✓
Power Amplifiers			✓
Microphone Power Supply			✓

APx Audio Analyzer *with* APx1701

Connections for tests requiring acoustic stimulus of the hearing aid with an APx1701 are shown in Figure 7. The analyzer's Unbalanced output Ch1 is connected to the APx1701 Power Amplifier Input 1 and Power Amplifier Output 1 is connected to the chamber loudspeaker. The test microphone is connected to the APx1701 Unbalanced Microphone Input 1, and Microphone Output 1 is connected to the analyzer's Analog Balanced Input Ch1.
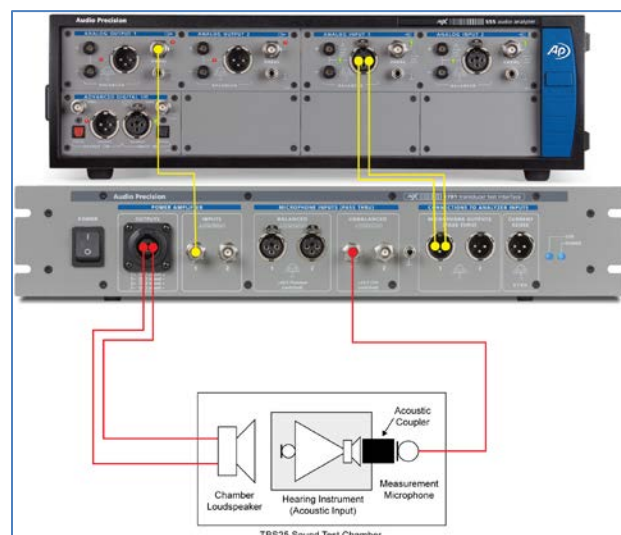
Figure 7. Connections for acoustic input to the hearing aid with APx1701.

Connections for tests of the hearing aid in telecoil mode are illustrated in Figure 8. In this case, Analog Unbalanced Ch2 of the audio analyzer and power amplifier 2 of the APx1701 are used to drive the induction loop through a series resistor. A cable to sense the voltage drop across the induction loop series resistor is connected to Analog Input Ch2 of the analyzer.

For the battery current measurement with an APx1701, a DC power supply is used with a series resistor, as shown in Figure 9. A cable to sense the voltage drop across the series resistor is connected directly to Balanced Analog Input Ch2 of the analyzer.
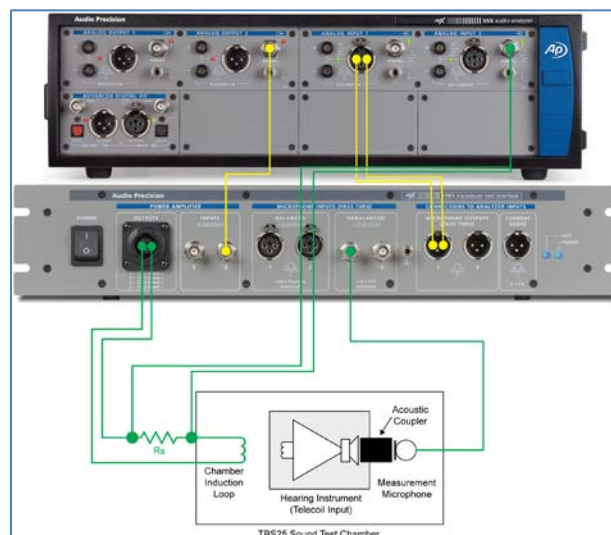

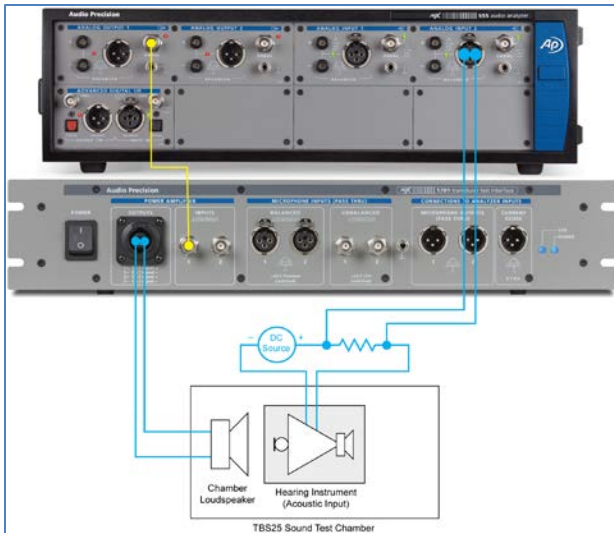Figure 8. Connections for the chamber induction loop with APx1701.

Figure 9. Connections for battery current measurement with APx1701.

APx Audio Analyzer *without* APx1701

When an APx audio analyzer is used without an APx1701, an external power amplifier is required to drive the chamber loudspeaker and an external microphone power supply is required for the test microphone. Connections for this signal path are shown in Figure 10. Note: The analog output connector used on the audio analyzer depends on whether the power amplifier has a balanced or unbalanced input. The project file assumes that a power amplifier with balanced input is used, but this can easily be changed to accommodate a power amplifier with unbalanced input.
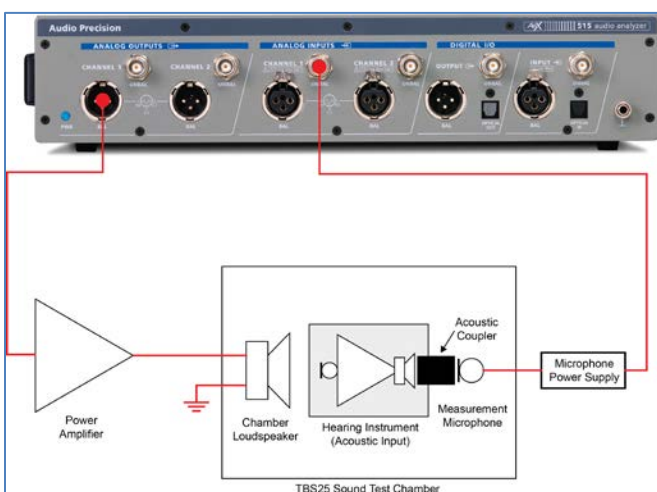
Figure 10. Connections for acoustic input to the hearing aid without APx1701.

An external power amplifier is also required to drive the induction loop. Figure 11 illustrates the induction loop connection for a power amplifier with a balanced output, which is assumed in the project file accompanying this Technote. Once again, the project file can easily be changed to accommodate a power amplifier with unbalanced input.
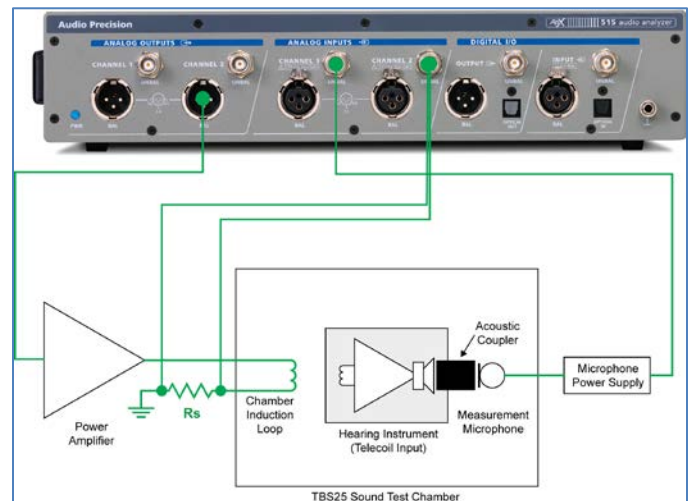

Figure 11. Connections for the chamber induction loop without APx1701.

To sense the voltage drop across the series resistor used in the battery simulator supply circuit, connect a cable to Analog Balanced Input Ch2 (as shown in Figure 9 for the case with APx1701).

APx Hearing Aid Plug-in

The project files accompanying this Technote use the APx Hearing Aid Plug-in. This software plug-in, available for download from ap.com, should be installed before using the project files. It adds the following results that are specific to hearing aid test.

1. HI – High Frequency Average (HFA)
2. HI – RTG Target Normalized
3. HI - Frequency Range:
4. HI – Equivalent Input Noise (EIN)
5. HI - Equivalent Test Loop Sensitivity (ETLS)
6. HI - HFA MASL
7. HI - Battery Current

Project File Structure

The three project files have essentially the same structure. They are intended to be used in Sequence Mode and they

include two sequences named Calibrate and Test DUT. The Sequence Control in the Navigator panel (Figure 12) is used to select which sequence is active.
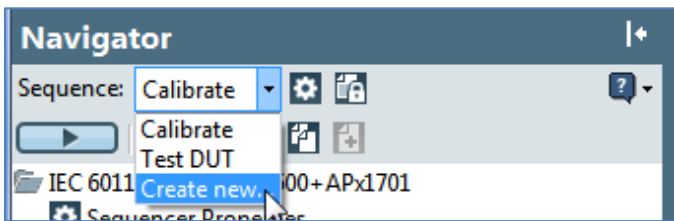
Figure 12. The Sequence control in the APx500 Navigator.

Calibrate Sequence

The Calibrate sequence does the following:

1. Prompt the operator to insert the test microphone in a sound level calibrator.

2. Sets the microphone sensitivity based on the calibrator measurement so that acoustic input levels are calibrated.

3. Uses the Auto-Set-Generator Level feature to calibrate the acoustic output sensitivity at 1.0 kHz.

4. Measures the frequency response at the test point and uses the inverted relative frequency response to equalize (EQ) or level the sound field such that the EQ'd response is flat at the test point.

5. Measures the ambient noise at the chamber's test point in terms of wideband (200 Hz to 8 kHz) level and 1/3-octave spectrum.

6. Measures the EQ'd frequency response at the test point compared to limits.

7. In analyzers other than APx511, measures and exports the DC offset of the input used to sense the battery current.

8. In analyzers other than APx511, uses the Auto-Set-Generator Level feature to set the generator dBrG for the reference level of 31.6 mA/m in the induction loop.

This leveling of the sound field is accomplished in two signal paths. In the first signal path, named "SP1: Cal Mic + Chamber", the frequency response is measured and the relative response result is exported via a Sequence Step to an Excel file whose name is specified by user-defined variables. The second signal path, named "SP2: Acoustic Measurements", has its Measurement Sequence Settings node configured to copy the references (including acoustic input

and output sensitivities) from the first signal path. Then, a Sequence Step is used in the Signal Path Setup measurement to import the relative response data, invert it and apply it as an Output EQ curve. A typical EQ'd frequency response resulting from this process for a TBS25 sound test chamber is shown in Figure 13, with the limits from IEC 60118-0 shown as red lines.

Test DUT Sequence

The Test DUT sequence in the accompanying project files uses three signal paths. Measurements involving acoustic stimulus of the hearing aid are in the signal path named "SP2: Acoustic Measurements". The battery current measurement is in a signal path of its own, and the induction loop measurements are in a signal path named "SP5: Telecoil Measurements".

The Device ID Prompt step in the Pre-Sequence Steps node of the Test DUT sequence is checked, so that the user will be prompted to enter the Device ID (e.g., serial number) when the sequence is run.

Acoustic Measurements Signal Path

The first measurement in the Test DUT sequence is an APx Frequency Response measurement which has been renamed to "OSPL90 Frequency Response (chirp)". This measurement uses a log-swept sine stimulus, also known as chirp. It is very fast (approximately ½ second duration as configured in these project files). However, chirp measurements typically require a relatively high signal to noise ratio (SNR). This chirp stimulus was chosen for the OSPL90 measurement because at 90 dBSPL, the SNR should be quite high.
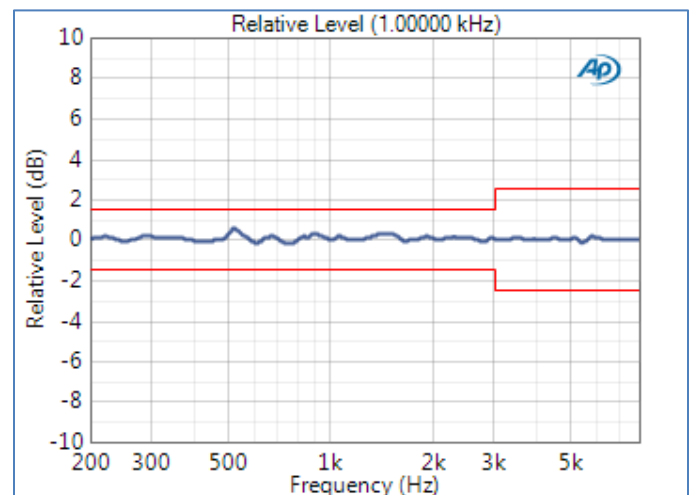

Figure 13. EQ'd frequency response at the test point compared to limits in IEC 60118-0.

Note that chirp measurements typically have high frequency resolution. For example, the RMS Level result in this measurement has almost 1000 points, linearly spaced from 200 Hz to 8 kHz. To reduce the number of points, a Specify Data Points derived result (renamed to OSPL90 Curve) has been used to obtain the frequency response at ISO preferred 1/12-octave frequencies. A resolution of perhaps 1/24-octave could similarly be specified.

The OSPL90 measurement has a result named HFA-OSPL90 which is an HFA result derived from the RMS Level result. The measurement also has a sequence step to export the HFA-OSPL90 result to a file specified by APx environment variable APPDATA and user-defined variables named "AppFolder" and "HFAOSPL90File" (Figure 14). This exported result will be used in a subsequent measurement to calculate the target for RTS.

The second measurement in the Test DUT sequence is an APx Multitone Analyzer measurement which has been renamed to "OSPL90 Frequency Response (Multitone)". This is included as an alternative for users who prefer a multitone stimulus to chirp. It uses a custom multitone signal with 1/12-octave spacing from 200 Hz to 8 kHz. Such custom multitones are easy to create in this measurement. Note that the generator level was set to 97.2 dBSPL for this measurement. This level setting was required to achieve an overall level of 90 dBSPL for this measurement, because the multitone signal has a much higher crest factor than a single sine tone.
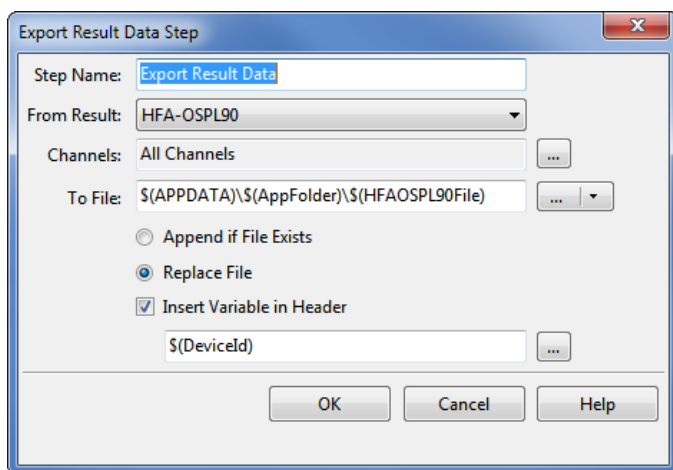

Figure 14. Sequence step to export HFA-OSPL90

The third measurement in the Test DUT sequence is an APx Bandpass Frequency Sweep measurement which has been renamed to "FOG Curve". As required, the generator level is set at 50 dBSPL. In this measurement, a custom sweep ta-

ble corresponding to the ISO preferred 1/12-octave frequencies from 200 Hz to 8 kHz was imported from an Excel file. The measurement uses a bandpass filter at each frequency step. In this case, it was set to "Window width" meaning the narrowest filter possible. In this measurement, the Gain result was renamed to Full-On Gain Curve, and an HFA result was derived from it and renamed HFA-FOG. A built-in derived result, renamed to FOG Maximum, was also used to find the maximum value of the full-on gain curve.

The fourth measurement in the Test DUT sequence is an APx Bandpass Frequency Sweep measurement which has been renamed to "Basic Frequency Response (1/12-Oct)". This measurement has a Sequence Step prompting the operator to set the hearing aid at RTS. This could be replaced by a Run External Program step configured to programmatically set the hearing aid at RTS. The Gain result in this measurement has been renamed to "Basic Gain Curve", and an HFA result derived from it has been renamed to "RTG".

The measurement also has an HI – RTG Target Normalized result derived from the basic gain curve. This result reads the HFA-OSPL90 result exported earlier and uses it to calculate the normalized RTG target. If the Match Device ID check box in this result is checked, unless the current Device ID matches the one in the HFA-OSPL90 file, it will display an error message on the graph without returning data.

The Basic Frequency Response measurement also has a Sequence Step to export the RTG result, so it can be imported and used for calculations in subsequent measurements. Like the step to export HFA-OSPL90, above, it also uses variables for the file specification and inserts the Device ID in the header.
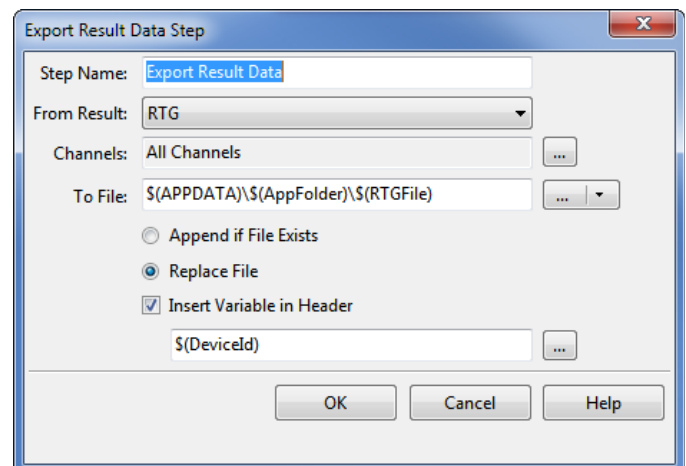

Figure 15. Sequence step to export the RTG result from the Basic Frequency Response measurement

The next four measurements in the Test DUT sequence are THD measurements as detailed in Table 2. These are instances of the APx THD+N measurement which have been renamed for clarity. All the primary results in these measurements have been deleted except THD Ratio, since this is the only result required in this case.
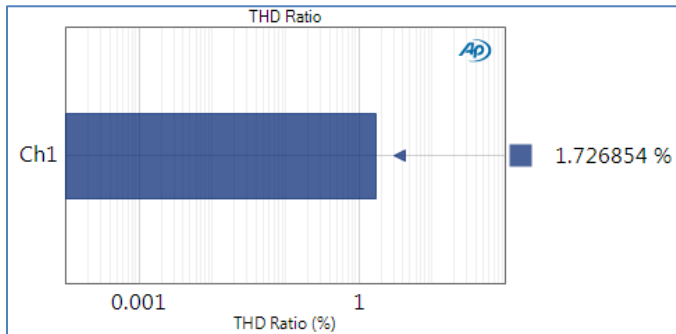

Figure 16. THD measurement at 500 Hz and 70 dBSPL

The ninth measurement in the Test DUT sequence is an APx Bandpass Frequency Sweep measurement which has been renamed to HFA at 50 dBSPL (for EIN). As the name implies, the purpose of this measurement is to measure the HFA-Gain at 50 dBSPL so that it can be used in the subsequent EIN measurement. This measurement is a sweep over the three HFA frequencies. It has a Sequence Step to export the HFA-Gain50 result, again using variables and with the Device Id inserted in the file header.

The tenth measurement in the Test DUT sequence is an APx Noise (RMS) measurement which has been renamed to Equivalent Input Noise. The measurement's high-pass and low-pass filters have been set to the required range of 200 Hz to 8 kHz. In addition, the acquisition time (accessible from the Advanced Settings button) has been set to 0.5 seconds, so that the noise measurement will be integrated over this time. In this measurement, the primary result Noise Level has been renamed to Output Noise Level. A second result, HI – Equivalent Input Noise (EIN), has been derived from the Output Noise Level. The EIN result uses variables to read the HFA-Gain50 result exported in the previous measurement and uses it to calculate EIN. A Match Device ID check box can be used to ensure that the current Device ID matches the one written to the header of the HFA-Gain50 file.

The next measurement in the Test DUT sequence is an APx Stepped Level Sweep measurement that has been renamed to "Steady State Input Output". As shown in Figure 5, it measures the output of the hearing instrument as the input SPL is increased from 50 dBSPL to 90 dBSPL in 5 dB

steps. If required, the measurement settings can easily be changed to increase the range of levels swept or decrease the size of the level steps.

Next in the sequence is an Attack and Release measurement. This measurement, originally named "Attack and Release (ANSI S3.22)" is part of the APx Attack and Release plug-in, which can be downloaded from ap.com. Figure 17 shows the acoustic output signal from a hearing aid as the SPL of a sinusoidal signal at 2 kHz is abruptly increased from 55 to 90 dBSPL and then decreased from 90 to 55 dBSPL. This measurement determines the attack time and release time of the hearing instrument as explained above. Two other variations of this measurement are included with the Attack and Release plug-in, if more flexibility is required.
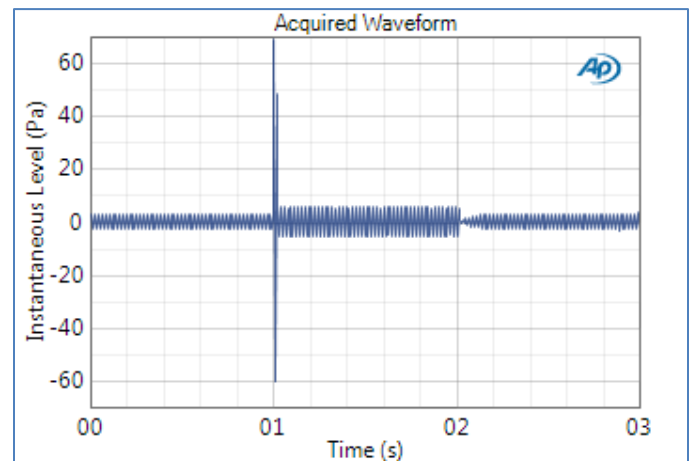

Figure 17. Sound pressure waveform measured during the attack and release test.

The next measurement in the Test DUT sequence is an APx measurement named "IMD Frequency Sweep (DFD)". This measurement is automatically renamed by the APx500 software to append the IMD Type in parentheses (in this case DFD). As required by IEC 60118-0, the difference frequency has been set to 125 Hz, with the higher frequency f_2 swept from 8 kHz to 350 Hz. In the project files, 31 points with logarithmic spacing was specified, but this can easily be changed as required. The primary result is the DFD Ratio plotted versus mean frequency, as shown in Figure 6, above.

The last measurement in the Acoustic Measurements signal path conducts a fractional octave analysis of a speech signal passed through the hearing aid. It is an instance of the Octave Analysis (by FFT) measurement that has been renamed to "Octave Analysis of Speech". The measurement

is part of the APx Octave Analysis plugin, available for download at ap.com. This measurement is not required by the IEC or ANSI standards noted above, but is included for demonstration purposes. Note that the APx generator is calibrated for sine signals. Therefore, when generating speech or other such signals with a relatively high crest factor, the generator level setting must be increased accordingly.
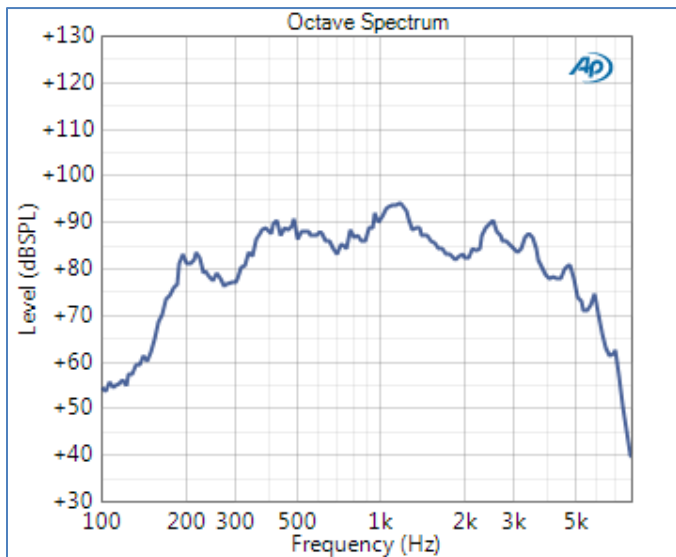

Figure 18. 1/24-octave spectrum of a speech signal processed by a hearing aid.

Battery Current Signal Path

In the Test DUT sequence, the signal path named "SP3: Battery Current" is used to measure the current draw of the hearing instrument battery.

In the case of the APx511, the Input Connector in this signal path is set to Battery Current. When this input is selected, a button labeled "Zero Adjust..." is visible in Signal Path Setup. This button can be used to zero out any dc offset present at the analyzer input, for improved accuracy. This requires turning off the battery supply voltage, which could cause the hearing instrument to lose any settings stored in volatile memory. Accordingly, a warning message is displayed. There is one measurement in this signal path: an APx DC Level measurement that has been renamed to Battery Current. On the APx511, the battery current will be measured directly and displayed.

In the two project files intended for APx audio analyzers other than the APx511, the Input Connector for the battery current signal path is set to Analog Balanced. In these project files, the battery current signal path also contains an

additional APx DC Level measurement that has been renamed to Measure DC Level Offset. This measurement contains a sequence step to export the DC Level result. Its check box is unchecked in the Test DUT signal path, but is checked in the Calibrate sequence. The intent is that during the Calibrate sequence, before the hearing aid battery supply is switched on, the DC Level offset can be measured and exported to a CSV file.

The other DC Level measurement in the battery current signal path, which has been renamed to Battery Current, has its check box checked in the Test DUT sequence. This measurement contains a result from the Hearing Aid Plugin named HI – Battery Current. This result is derived from DC Level. It calculates the current by simply dividing the measured dc voltage by the resistance value entered in its Series Resistance control beneath the graph. The result also has a check box labeled Compensate DC Offset. If checked, this result reads the dc offset exported during the Calibrate sequence and subtracts if from the measured DC Level before calculating the current. APx variables are used to specify the name of the dc offset file in this result and in the sequence step above, which creates the file.

Telecoil Measurement Signal Path

In the Test DUT sequence, the signal path named "SP5: Telecoil Measurements" is used for the hearing aid Telecoil mode measurements with an induction loop. When the APx511 is used, the Output Connector in this signal path is simply set to Telecoil. This selects the current mode induction loop amplifier, and changes the units of the Generator Level control in all measurements in the signal path to Amperes.

When an analyzer other than the APx511 is used, the Output Connector for the Telecoil Measurements signal path is:

- Transducer Interface, when an APx1701 is used, or

- One of the analog outputs (Balanced or Unbalanced) when a power amplifier is used.

In both the above cases, the signal path's Measurement Sequence Settings node is configured to copy the References from the signal path named SP4: Set Telecoil Reference (31.6 mA/m). This signal path is run during the Calibrate sequence. Its one measurement is Signal Path Setup, which is configured to run the Auto Set Generator Level procedure. This procedure regulates the generator output voltage to find the level required to achieve the specified target value. The project files supplied were configured for a resistor of 200 Ω in series with the induction loop of a TBS25

sound test chamber. Hence, a target value of 1.580 Vrms was specified. This level corresponds to the voltage drop across a 200 Ω resistor when the current flow through it is 7.9 mA (the current needed to achieve 31.6 mA/m in the TBS25 chamber's induction loop). If a different series resistor is used (or a different induction loop), the Target Value should be changed accordingly. The settings for this procedure are accessible from the button labeled "Auto Gen Level…" in the References Panel of Signal Path Setup. When the Auto-Set Generator Level procedure is finished, it automatically sets the dBrG reference to the value needed to achieve the Target Level; i.e., setting the generator level to 0 dBrG will generate a current of 7.9 mA, which creates a magnetic field strength of 31.6 mA/m in the TBS25 test chamber. This dBrG reference is one of the references copied to the Telecoil Measurement signal path when the Signal Path Setup measurement is run.

The Telecoil Measurements signal path contains an APx Bandpass Frequency Sweep measurement which has been renamed to "SPLIV Frequency Response". A Bandpass Frequency Sweep is recommended for induction loop measurements, which are inherently noisy, because it measures the bandpass-filtered response at each frequency, rather than the wideband level, like Stepped Frequency Sweep.
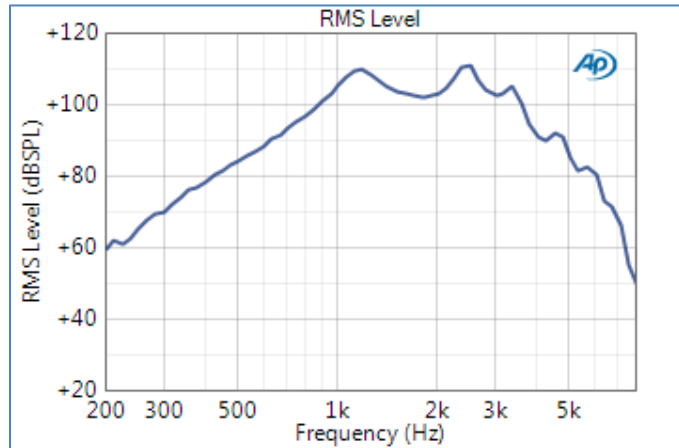

Figure 19. SPLIV frequency response of a hearing aid.

The SPLIV Frequency Response measurement contains an HFA result derived from the RMS Level result named HFA-SPLIV. This is one of the metrics specified in IEC 60118-0. Note that to meet the standard, this sweep need only be conducted at the three HFA frequencies, so that the HFA-SPLIV can be calculated. The sweep in the project files is specified at ISO 1/12-octave frequencies, in case users want more detailed frequency response information.

The SPLIV Frequency Response measurement also contains the HI – Equivalent Test Loop Sensitivity (ETLS) result, which is derived from the HFA-SPLIV result. It reads the RTG data exported previously from the Basic Frequency Response measurement in the Acoustic Measurements signal path. Like other results in the Hearing Aid plugin, this result uses variables, and optionally checks that the Device ID written to the RTG file matches the current Device ID.

The last measurement in the Telecoil Measurements signal path is an APx Bandpass Frequency Sweep measurement, which has been renamed to "HFA MASL (10 mA/m FOG)". This measurement has a prompt instructing the operator to set the hearing instrument at full-on gain. It sweeps the three HFA frequencies with the generator level set to −10 dBrG, which corresponds to 10 mA/m. This measurement contains two derived results from the Hearing Instrument plugin: HFA and HFA MASL. HFA MASL has a control below the graph to specify the magnetic field strength in mA/m. It calculates the HFA MASL from the HFA Level, as described above.

In these project files, the Report check box at the bottom of the Navigator is checked. As such, when the Test DUT sequence is complete a detailed test report of the entire sequence of measurements will be displayed. This report can be saved in one of the several formats supported.
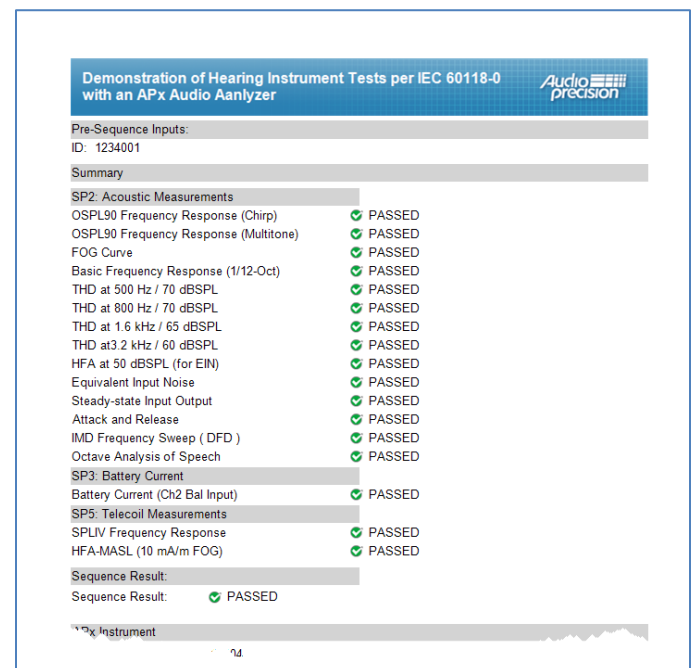

Figure 20. Summary section of APx test report.

Conclusion

This concludes the discussion describing methods of conducting the primary hearing aid measurements specified in IEC 60118-0 and ANSI S3.22 with APx500 series audio analyzers. The extensive feature set of the APx platform can be used to conduct many more types of audio measurements, if required.

The APx project files described in this technote are available for download at ap.com.