

Verifit & RM500SL Test Signals & Analysis

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1 Introduction

The Verifit and RM500SL are full-duplex dual-channel audio measurement systems designed for the testing and fitting of all types of hearing instruments and many assistive listening devices. They provide an acoustic test chamber for the testing of devices coupled to standard 2 cm³ couplers as well as sound-field speaker(s) for on-ear measurement of hearing aid performance. The Verifit test chamber contains two loudspeakers for the functional testing of directional hearing instruments. Test signals are delivered to the device under test via the test chamber loudspeaker(s), the test chamber telephone magnetic-field simulator (TMFS), the test chamber magnetic field test loop (Verifit only) or the sound-field loudspeaker(s). Device output signals are measured in the 2 cm³ coupler via the coupler microphone or in a real ear via the probe microphone. Data from the 2 cm³ coupler may be viewed as coupler SPL, coupler gain or as estimated SPL in the ear canal (simulated real-ear measurements, S-REM). Data from the probe microphone may be viewed as ear canal SPL, as ear canal SPL re normal hearing threshold (i.e. dB HL), as insertion gain or as ear canal SPL in the context of an auditory area (Speechmap®).

2 The test signals

Sinusoidal, pseudo-random noise and digitized real speech signals are provided. Two signals derived from real speech, the International Speech Test Signal (ISTS) and the single-talker International Collegium of Rehabilitative Audiology (ICRA) signal, are also provided. Test signals are generated in real time by the digital signal processor (DSP) or played from 16 bit binary audio files stored in the on-board flash memory. In the Verifit, these audio files are up-loaded from the internal compact disc (CD) drive at power up. In order to provide a repeatable acoustic signal to the device under test, a reference microphone is used in conjunction with a digital control loop to maintain the desired band level at each frequency. Live audio (speech, music etc.) may also be used as a test signal but it is not controlled for spectrum or level.

Some useful facts about broad-band signals

Overall SPL is the SPL in a band containing all significant frequency components of the signal.

Spectrum level is the SPL in a band 1 Hz wide.

Band SPL is the SPL in a restricted frequency range. If the signal is uniform in the band, band SPL = Spectrum level + 10*log(bandwidth).

A spectrum is the band SPL, or spectrum level, in a series of adjacent bands.

For a broad-band signal, the overall SPL is greater than the band SPL and the band SPL is greater than the spectrum level. For a pure tone, the overall SPL, the band SPL and the spectrum level are the same.

A white noise signal has a spectrum level that is independent of frequency (i.e. constant SPL per Hertz).

A pink noise signal has a spectrum level that is inversely proportional to frequency (i.e. constant SPL per octave).

A fractional octave band (e.g. 1/3 octave) has a bandwidth that is proportional to frequency.

A pink noise signal has a flat spectrum when analyzed in fractional-octave bands.

A critical band is a band within which the loudness of a continuously-distributed signal of constant SPL is independent of bandwidth.

2.1 Narrow-band signals

2.1.1 Sinusoidal (pure tones)

Sinusoidal signals are used in the ANSI hearing aid tests, in Multicurve 2 cm³ frequency response tests (gain or output), in 2 cm³ and on-ear single frequency (manual) tests, in Insertion gain tests and in tone-burst maximum output (MPO) tests. The MPO stimulus consists of a series 128ms tone bursts with 128ms gaps at an SPL of 90 dB SPL in the test box and 85 dB in the sound field. Frequencies used are 1/12 octave except for the MPO test which uses 1/3 octave frequencies. Levels from 40 to 90 dB SPL are available at the reference microphone. Control of the level at each frequency is maintained by measuring the frequency response of the signal path to the reference microphone 256 ms before each test using a 256 ms, 50 – 55 dB SPL, multi-tone complex or a 256 ms chirp, depending on the test to follow. Drive levels at each frequency are then set to achieve the desired band levels at the reference microphone. For on-ear measurements, tones are frequency modulated (triangle, ± 3% over 128 ms).

2.2 Broad-band signals

2.2.1 Pink Noise signal

The pink noise signal is available for Multicurve 2 cm³ coupler gain and output frequency response measurements and for Insertion gain and Speechmap on-ear measurements. It is a pseudo-random signal composed of 1024 simultaneous tones summed to provide a crest factor of 12 dB. The spectrum of the pink noise signal is controlled by the reference microphone in conjunction with a digital feedback loop with a frequency resolution of 1/12 octave and a response time of about 1/3 of a second. Pink noise was selected as a test signal because it has equal energy per octave, producing a flat spectrum when analyzed in 1/12 or 1/3 octave bands. Figure 1 shows the 1/3 octave spectrum of the pink noise signal and the noise signal specified in ANSI S3.42-1997. Note that the ANSI S3.42-1997 spectrum represents speech peaks not the long-term average.

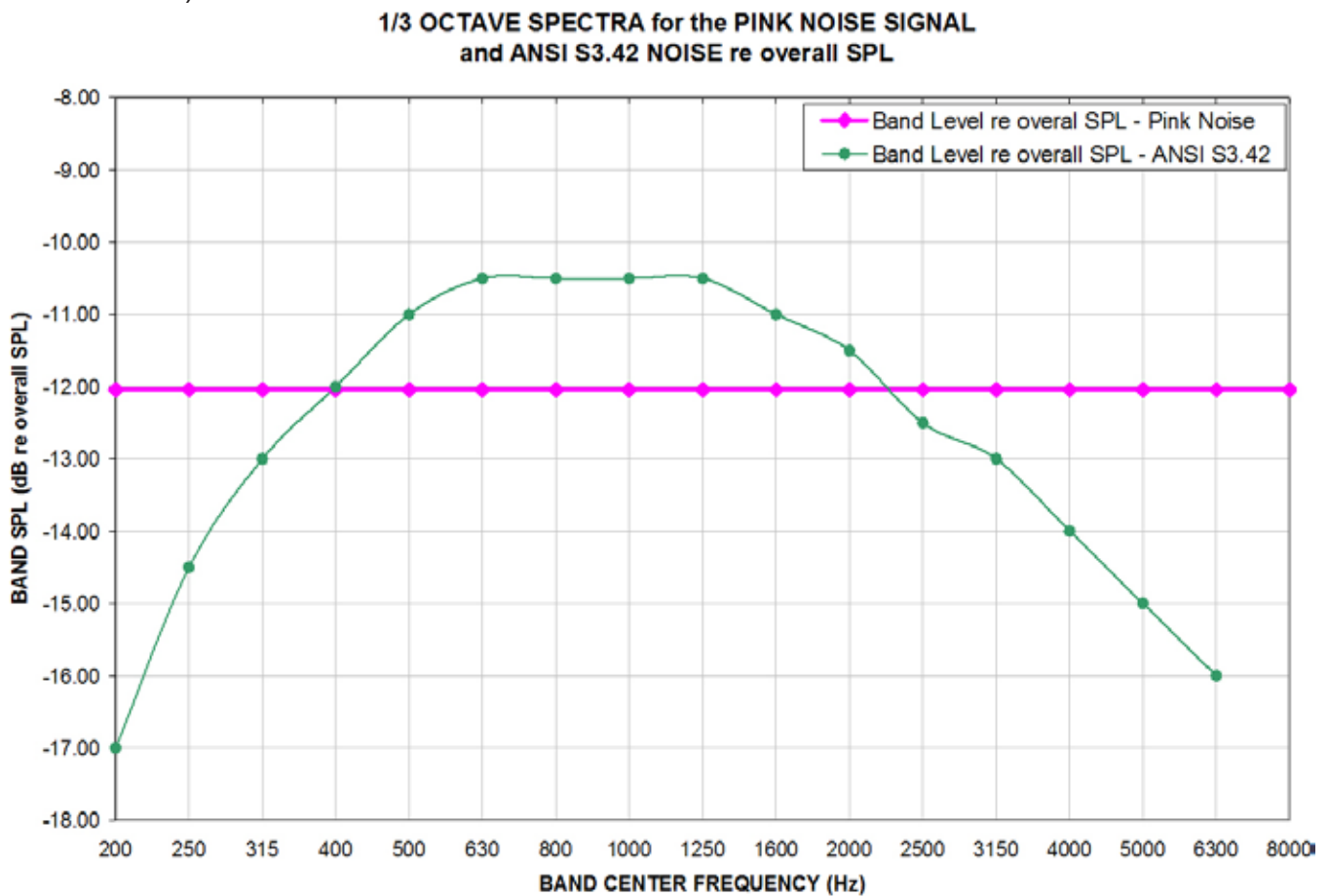
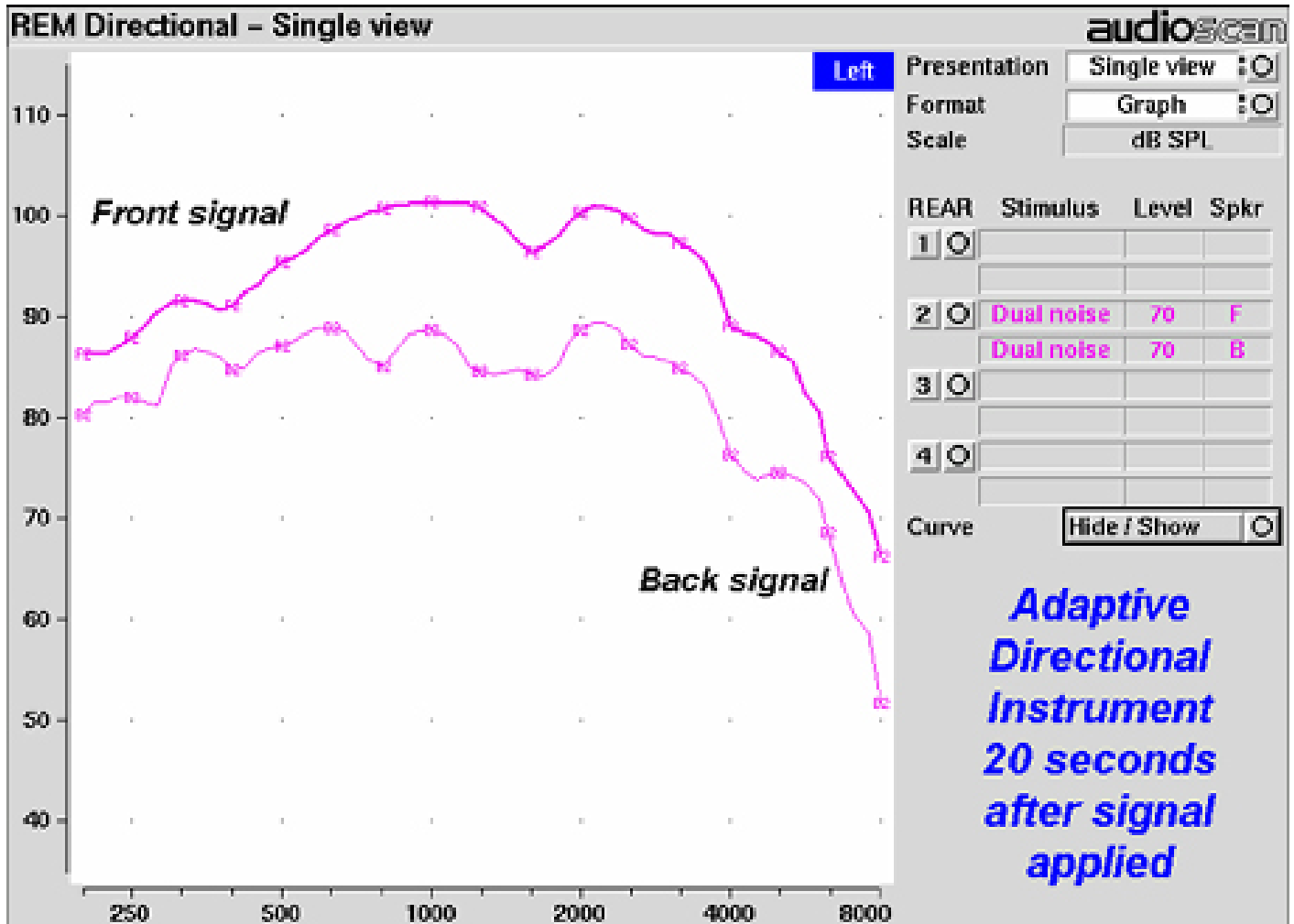


Figure 1: 1/3 octave spectra for the pink noise signal and the noise signal specified in ANSI S3.42-1997.

2.2.2 Dual-direction pink noise signal (Verifit only)

This unique test signal is available in the dual-source Verifit test chamber and for on-ear measurements by adding an additional sound-field speaker. It is similar to the pink noise signal described previously except that the component tones are presented simultaneously from two speakers, half from each. The spectrum of the tone complex from each speaker is independently controlled with 1/12 octave resolution

at the reference microphone. Two simultaneous response curves – one for each speaker – are derived from the measured output of the hearing aid. This provides a real-time measure of the functioning of the directional features of hearing aids that is independent of compression or noise reduction algorithms, unlike other methods that sequentially measure response from different directions and work only with these features disabled.



2.2.3 Real-speech signals

Real-speech signals are provided in Speechmap for both On-ear and Test box measurement modes. Four different digitized speech passages (2 male, 1 female and 1 child) are provided as well as the International Speech Test Signal (ISTS) and the single-talker International Collegium of Rehabilitative Audiology (ICRA) distorted speech signal. In addition, one of the speech passages has been filtered to provide 3 variations for evaluating frequency-lowering hearing aids. Each has a duration of 10 – 15 seconds and may be presented as a single passage or in a continuous loop. In order to provide a repeatable speech signal to the device under test, the signal path must be equalized prior to the presentation of the speech signal. This is accomplished by presenting a 896 ms pink noise burst at the selected speech level, 256 ms prior to the start of each speech passage and adjusting a digital filter to provide a flat response at the reference microphone.

The two “Speech-std” signals are by the same male talker, filtered to provide the long-term average speech spectrum (LTASS) recommended by Cox & Moore (1988) for average vocal effort. This is the LTASS assumed in the Desired Sensation Level (DSL) method of hearing aid fitting. In addition, four special versions of the Speech-std (1) test stimulus are provided in Speechmap to assist in the adjustment of frequency-lowering hearing aids. These are called Speech3150, Speech4000, Speech5000 and Speech6300. The Speech3150 stimulus has had the bands at 1000 Hz and above attenuated by 30 dB except for the 1/3 octave band at 3150 Hz which is unattenuated. Similarly, the Speech4000 stimulus has an unattenuated band at 4000 Hz, the Speech5000 stimulus has an unattenuated band at 5000 Hz and the Speech6300 stimulus has an unattenuated band at 6300 Hz (Figure 4). For these stimuli, the indicated level is for the unfiltered Speech-std (1) passage and the band levels in all unattenuated bands are the band levels present in the Speech-std (1) passage at that level. These three signals may be used to determine the amount of frequency shift provided by frequency-lowering hearing aids and the sensation level of the lowered components of the speech signal.

The “female” and “child” signals are presented “as recorded” without any processing and have been chosen to provide a range of spectra.

The ISTS was developed under the European Hearing Instrument Manufacturers Association (EHIMA) which holds the copyright. The sound file is available free of charge from the EHIMA website. The ISTS consists of 500 ms segments from recordings of 6 female talkers reading the same passage in American English, Arabic, Chinese, French, German and Spanish. These segments have been spliced together with appropriate pauses and filtered to match the average female spectrum from Byrne et al, An international comparison of long-term average speech spectra. *J. Acoust. Soc. Am.* 96 (1994), 2108-2120. For more details, consult the EHIMA website (www.ehima.com).

The ICRA distorted speech signal is a recording of an English-speaking talker that has been digitally modified to make the speech largely unintelligible. The resultant signal has many of the properties of real speech but has a harsh sound and lacks harmonic structure. The latter may be significant for hearing aids which use this property of speech to control noise reduction schemes. The LTASS is similar to the “Speech-std” signals up to 5 kHz.

Overall SPLs of 50, 55, 60, 65, 70 and 75 dB (at the reference microphone) are available. Soft and Average levels (50 - 70 dB SPL) have the same spectrum. A Loud vocal effort filter is applied to the 75 dB level signals. This is shown in Figure 5.

In the Test box mode in Speechmap (previously called S-REM) measurements made in the 2 cc coupler in the test chamber are converted to estimated real ear SPL. In this mode, the effects of hearing aid microphone location need to be included in the test signals so spectra are further altered to include the microphone location effects shown in Figure 6.

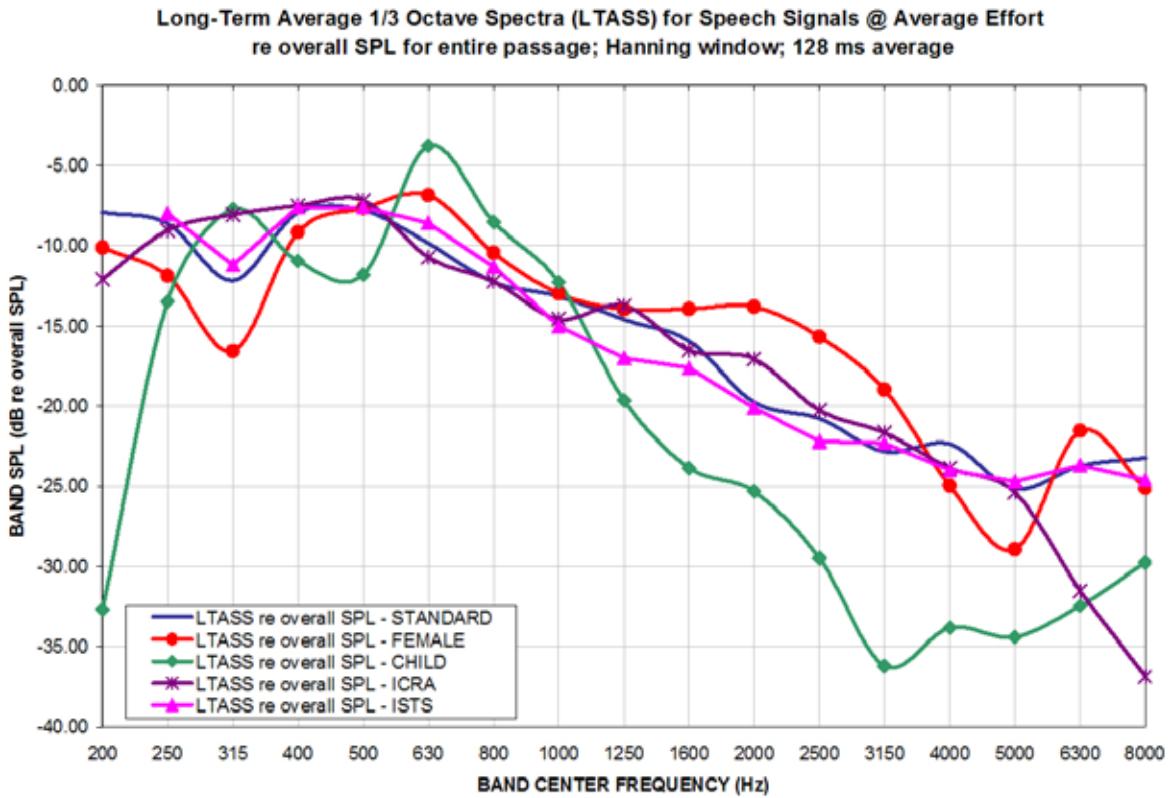


Figure 3: LTASS for Speechmap speech signals at average vocal effort

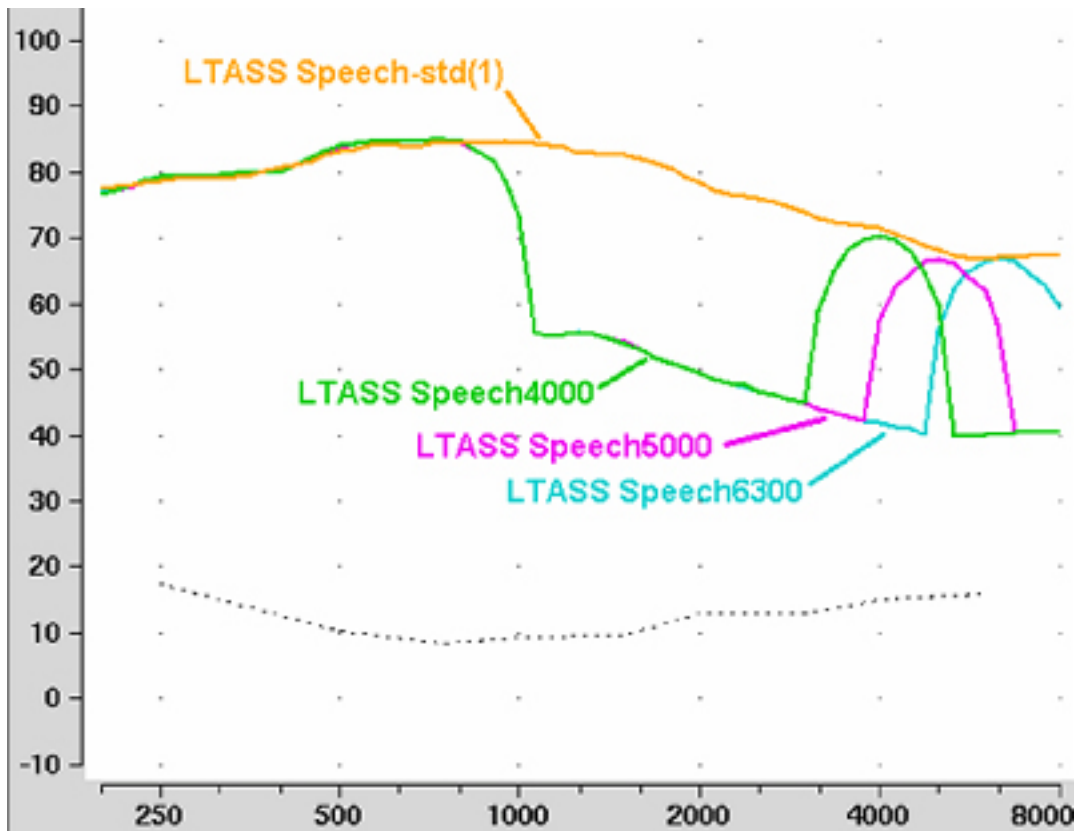


Figure 4: LTASS for Speech-std(1), Speech4000, Speech5000, Speech6300 for the FM boom location effects of Figure 6. Note that curves are 1/3 octave band levels at 1/12 octave intervals which causes the 1/3 octave bands at 4000, 5000 and 6300 Hz to appear broader than 1/3 octave. For clarity, the Speech3150 curve has been omitted.

LOUD VOCAL EFFORT EFFECT re nominal band SPL

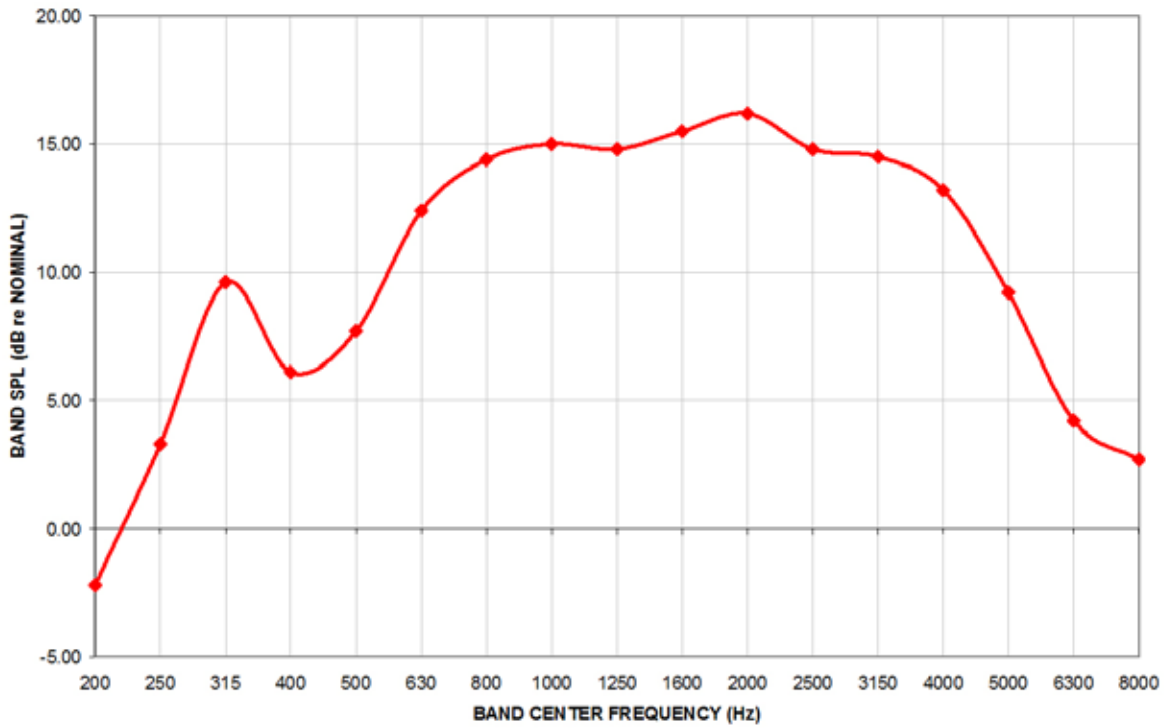


Figure 5: Loud vocal effort effect in dB re nominal band levels. Nominal band levels are band levels for an overall SPL of 65 dB. This shaping is applied to the 75 dB SPL speech and ICRA signals.

MICROPHONE LOCATION EFFECTS re nominal band SPL

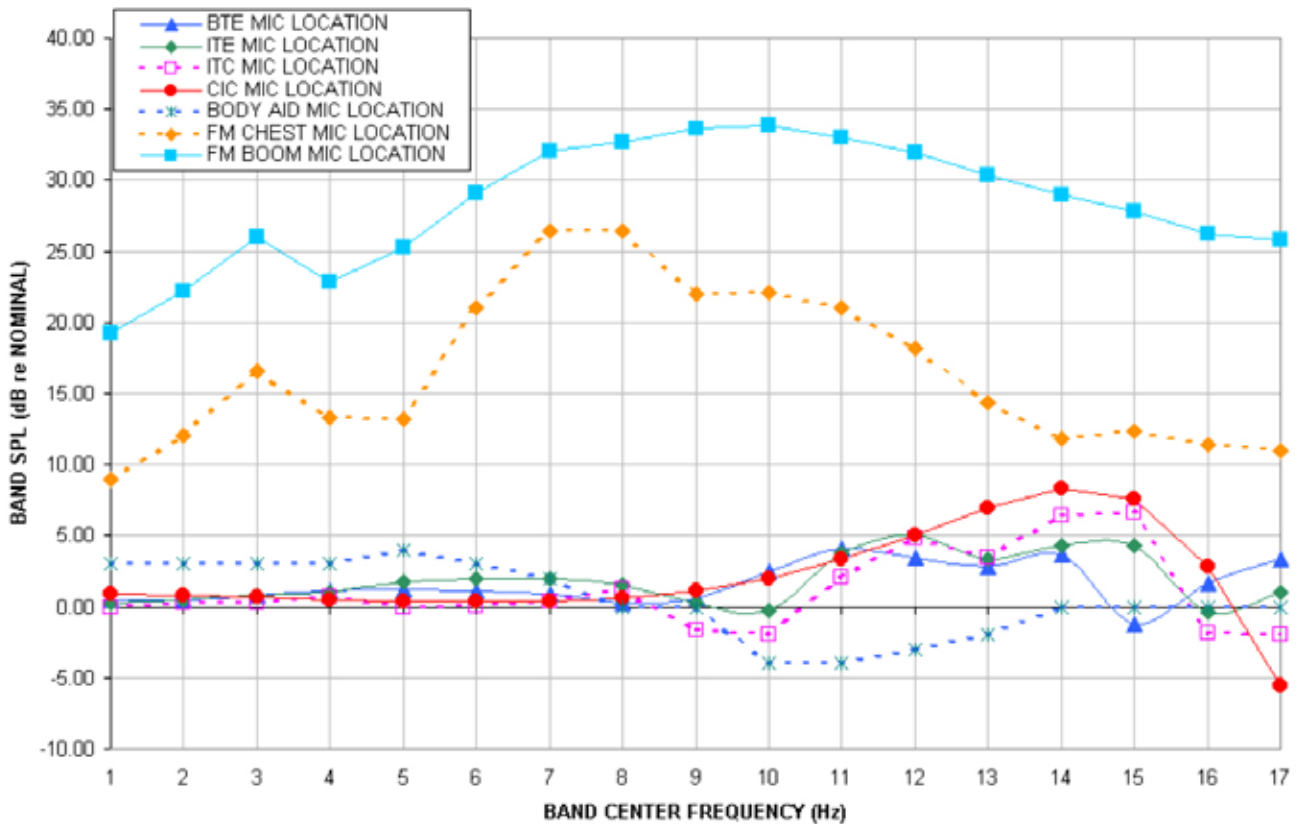


Figure 6: Microphone location effects in dB re nominal band SPL. Nominal band levels are band levels for an overall rms level of 65 dB SPL and average vocal effort in a free sound field.

In addition to these well-controlled and repeatable signals, live speech may be used as a test signal. It will, of course, be unequalized and at an uncontrolled level.

3 Analysis of broad-band signals

Broad-band signals contain energy at many frequencies simultaneously. Such signals are usually analyzed in a series of narrow frequency bands to produce a spectrum. The auditory system functions on a logarithmic frequency scale and analyzes broad-band signals in critical bands which approximate 1/3 octave bands (Figure 7). Using 1/3 octave bands for analysis of broad-band signals allows measured levels to be compared more readily to narrow-band behavioral measures, such as threshold. For this reason, analysis of broad-band signals in all On-ear measurements (including Speechmap Test box mode) is in 1/3 octave

bands. Other analyzers use narrower analysis bands, sometimes having constant bandwidth. As shown in Figure 7, this can result in significantly underestimating the audibility (or comfort or discomfort) of a complex signal.

It should be noted that, in all On-ear measurements, 1/3 octave band SPL is displayed (and reported when saving data to a file) at 1/12 octave frequencies which provides curve smoothing and causes the spectrum of narrow-band signals to appear broadened. When calculating overall rms from these data, the SPL at 225 Hz and subsequent 1/3 octave increments should be used.

Analysis of broad-band signals in all Test box tests, except Speechmap, is in 1/12 octave bands and the band SPL is displayed at 1/12 octave frequencies.

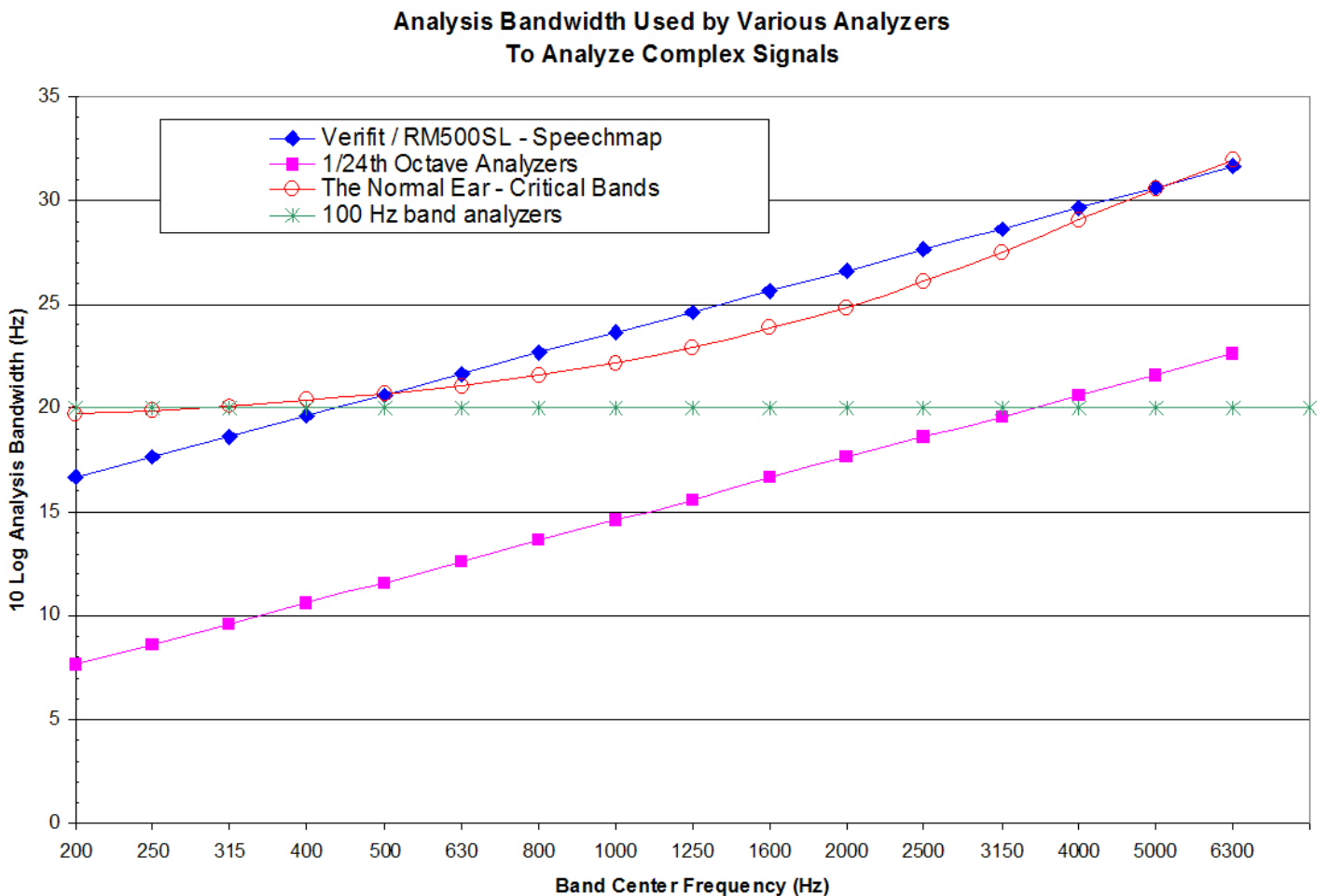


Figure 7: Analysis bandwidths of some real-ear analyzers and critical bands for a normal ear. If the analysis bandwidth is less than a critical band, aided response curves shown to be at threshold (or UCL) will actually be well above it.

3.1 Pink noise signal analysis

All band levels are calculated with an averaging time of 128 ms.

For a linear hearing aid, a pink noise stimulus will result in a response curve that has the same shape as one obtained using a swept pure tone. However, it must be remembered that, while a swept tone has the same band SPL as overall SPL, the band SPL for a noise signal is significantly lower than the overall SPL. Consequently, for a linear aid, output curves obtained using 1/12 octave analysis will be about 18 dB lower than the output curves obtained using a swept tone at the same overall SPL. As long as the hearing aid is linear, the gain obtained will be the same for both signals. Figure 8 shows output (A) and gain (B) for a linear hearing aid, obtained using a swept tone (1) and pink noise (2) with a 60 dB overall SPL.

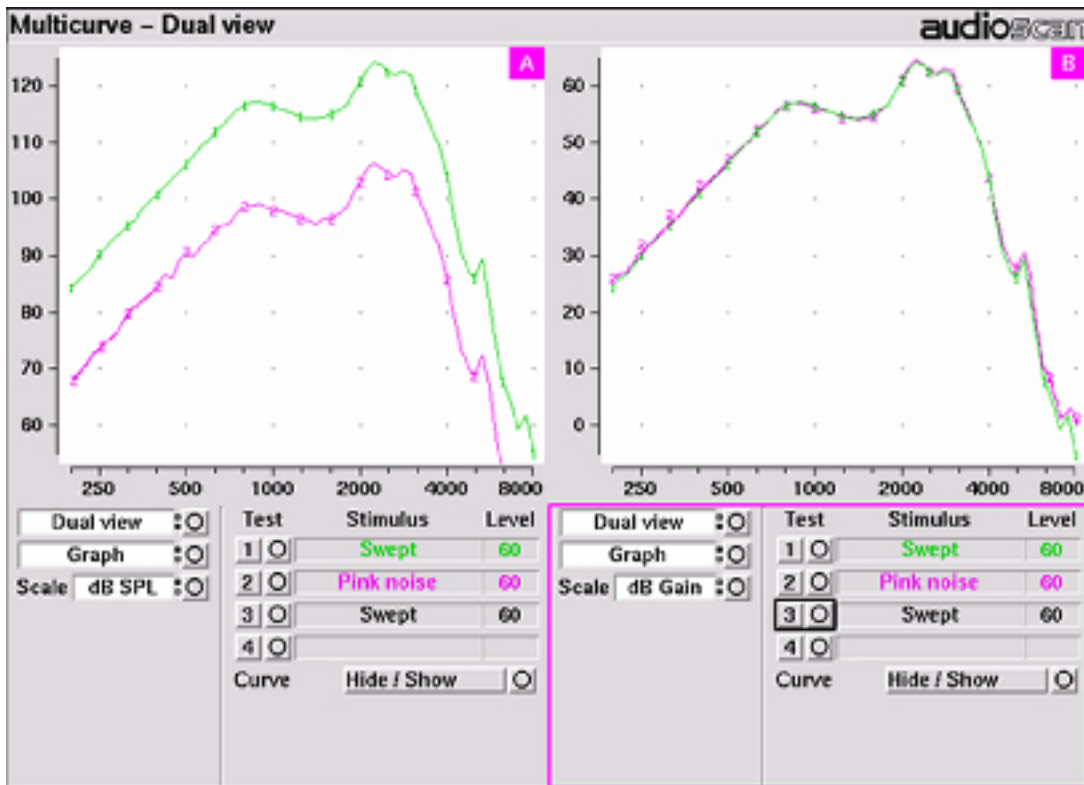


Figure 8: Output (left) and Gain (right) curves for a linear hearing aid, generated using a swept tone (1) and pink noise analyzed in 1/12 octave bands (2). (Dual view not available in RM500SL)

3.2 Real-speech signal analysis

One of the most-used measures of a speech signal is the long-term average speech spectrum (LTASS). This is a 1/3 octave spectrum averaged over a sufficiently long portion of the speech material to provide a stable curve. In practice a 10 second average meets this

requirement and, for this reason, all Speechmap passages are at least 10 seconds long. The LTASS curves displayed in Speechmap are 1/3 octave band levels at 1/12 octave intervals. It should be noted that this results in smoothing and the apparent broadening of the spectrum of a narrow-band signal. For example a 1/3 octave band of speech will exhibit a spectrum 2/3 octave wide.

The dynamic nature of speech is often characterized by the distribution of short-term levels in each 1/3 octave band. Historically, time periods of 120, 125 or 128 ms have been used. In Speechmap, 1/3 octave band levels at 1/12 octave intervals are derived every 128 ms. The level in each band that is exceeded by 1% of the samples (called L1, or 99th percentile) has historically been referred to as the speech peak for that band. The curve for these L1 levels is approximately 12 dB above the

LTASS. The level in each band that is exceeded by 70% of the samples (called L70, or 30th percentile) has historically been called the valley of speech for that band. The curve for these L70 levels is approximately 18 dB below the LTASS. The region between these two curves is often called the speech region, speech envelope or speech "banana". The speech envelope, when derived in this way, has significance in terms of both speech detection and speech understanding.

Generally, speech will be detectable if the L1 level is at or near threshold. The Speech Intelligibility Index (SII) is maximized when the entire speech envelope (idealized as a 30 dB range) is above (masked) threshold. This will not be an SII of 100% (or 1) because of loudness distortion factors, but higher SII values will not produce significantly higher scores on most test material.

The speech-reception threshold (SRT) is attained when the LTASS is at threshold (approximately - depending on test material and the individual). These scenarios are shown in Figures 10 - 12 which follow.

It should be noted that analysis methods which use shorter time periods produce higher peak levels and significantly different speech envelopes. In order to produce results that can be directly compared to measures of threshold (and UCL), the analysis time period needs to approximate the integration time of the ear. Although this varies with frequency and individuals, a value between 100 - 200 ms is likely. The Verifit and RM500SL use a 128 ms analysis time period as an approximation because it also has considerable historic support.

Because the spectrum of a speech signal varies with time, it is necessary to average measurements over

several seconds to obtain a stable, repeatable result. However, the effects of adjustments to hearing aid parameters need to be seen quickly in order to be useful for optimization. To resolve these conflicting needs, speech signals are first presented in a repeating loop with a sound-field equalization test prior to the start of each passage. A running calculation of the short-term speech envelope and average is performed on 2 - 3 seconds worth of data and is displayed in real time so that the effects of adjustments to hearing aid parameters are easily viewed. Pressing the Continue key causes the passage to restart and run in its entirety. The envelope and average are calculated over the full passage to provide stable and repeatable data.

When using live speech as a test signal, a "Freeze curve" function is available to capture the short-term spectra for examination and counseling purposes.

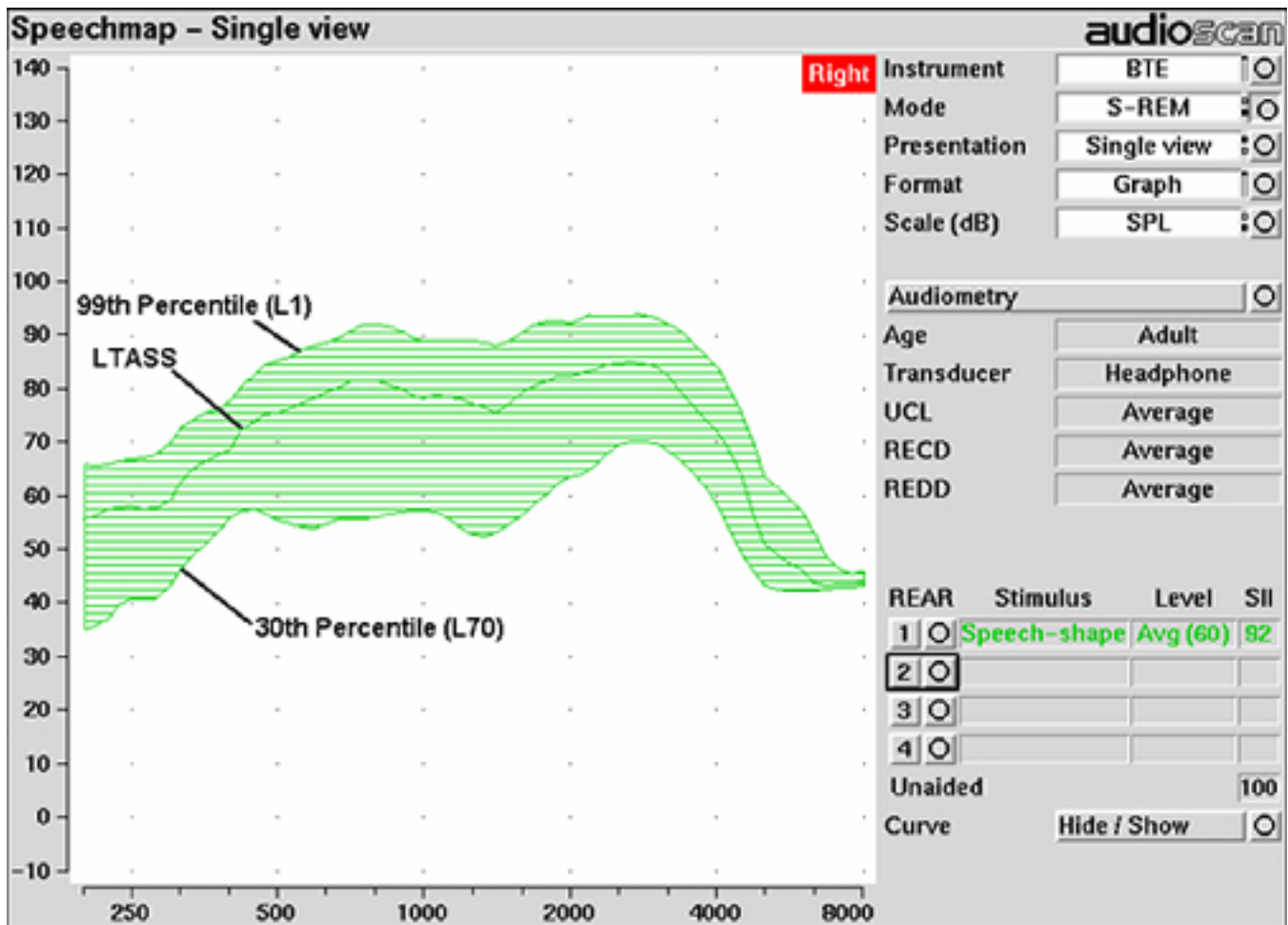


Figure 9: Example of speech envelope and LTASS

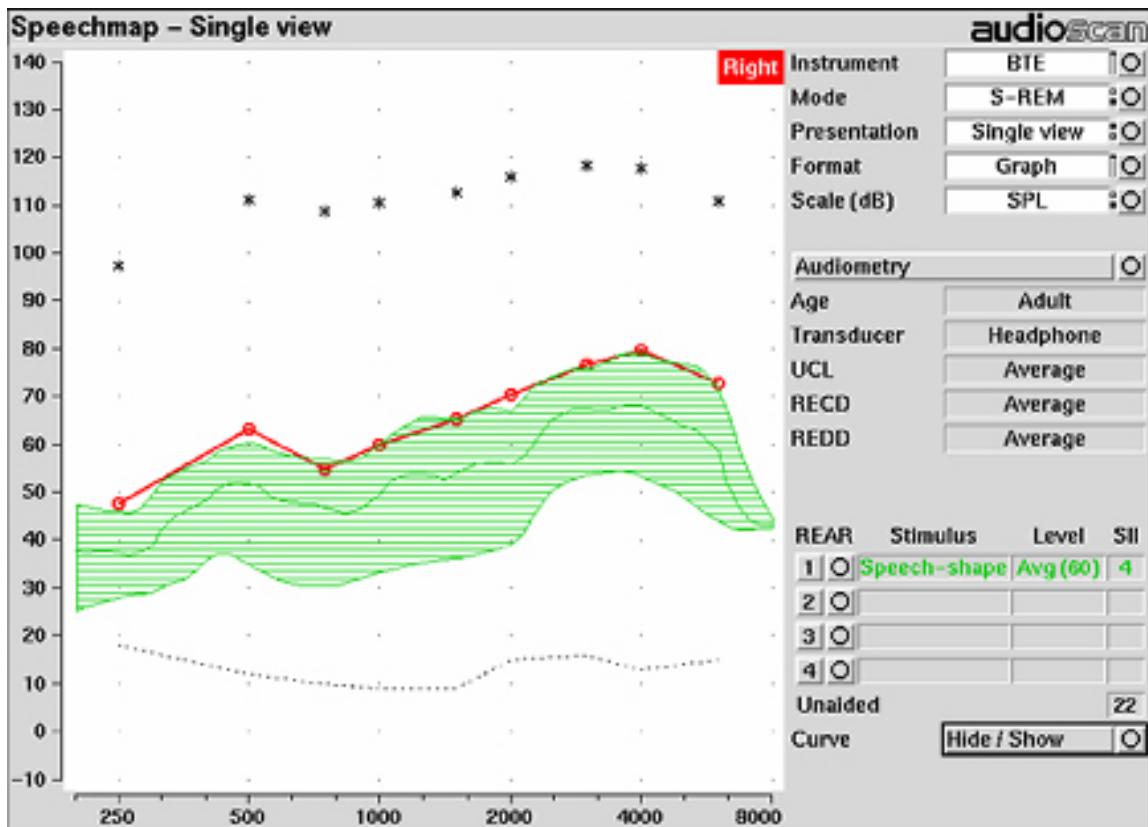


Figure 10: Speech is detectable but not understandable if the 99th percentile is at threshold

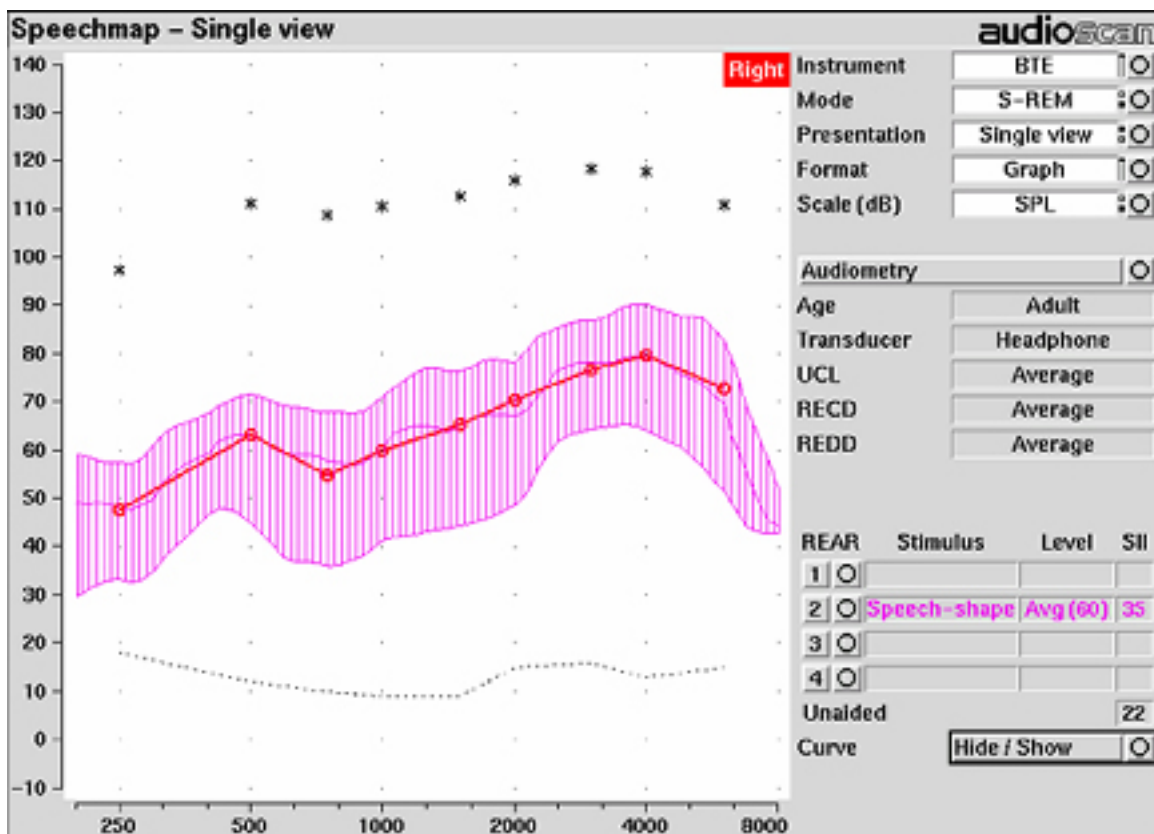


Figure 11: When LTASS is at threshold, SII \approx 33% which corresponds approximately to SRT

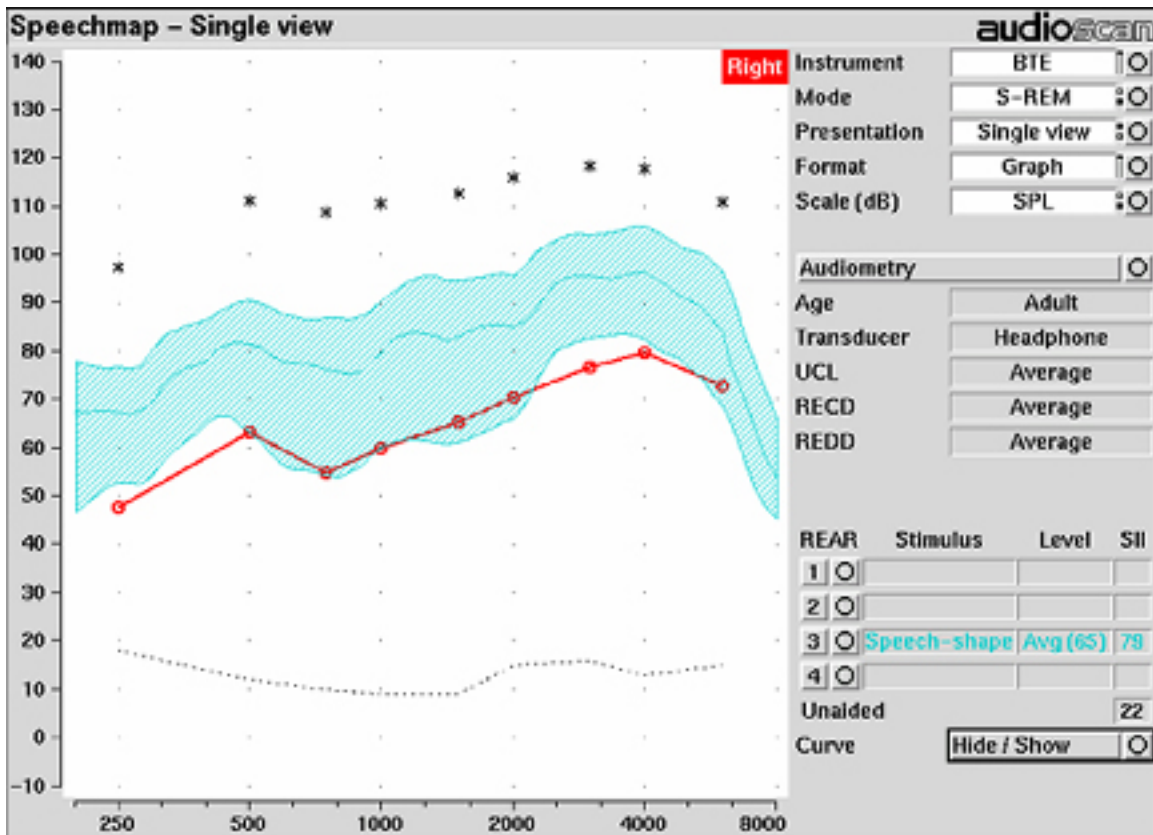


Figure 12: The highest SII is obtained when the entire speech envelope is above threshold

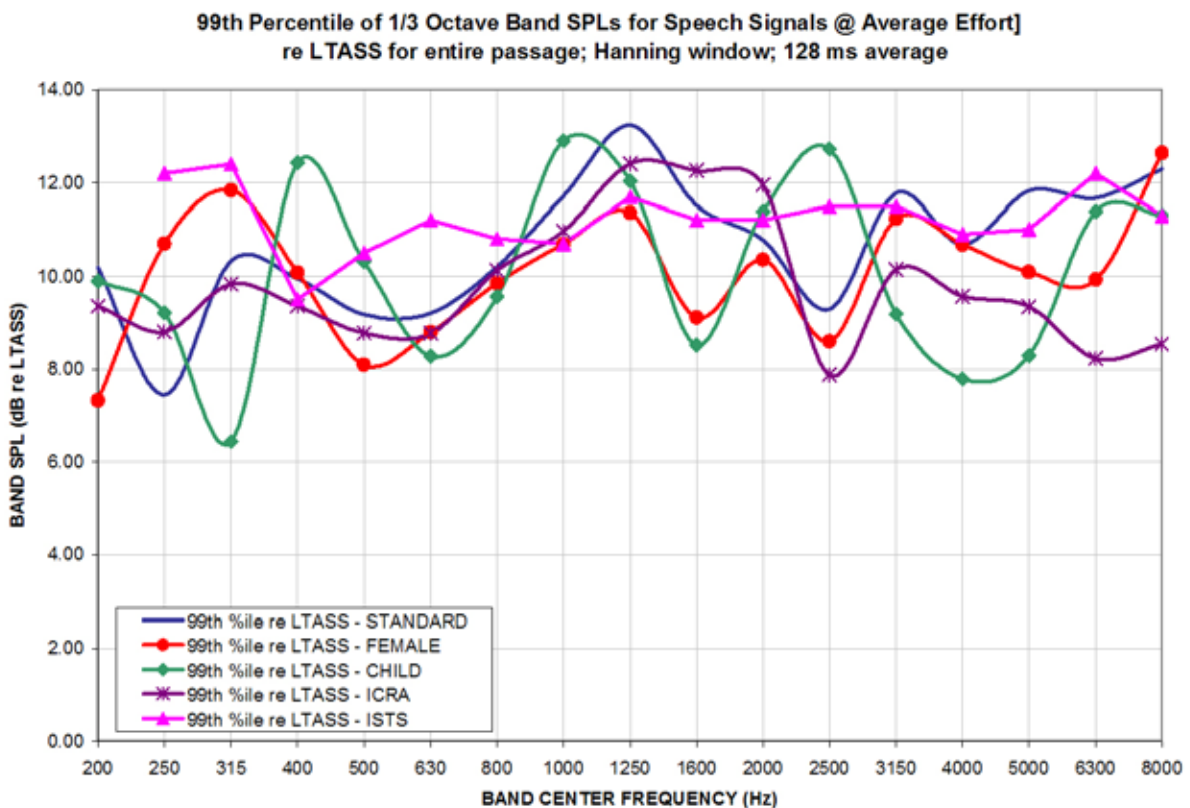


Figure 13: Level of 99th percentile for Speechmap speech signals at average vocal effort

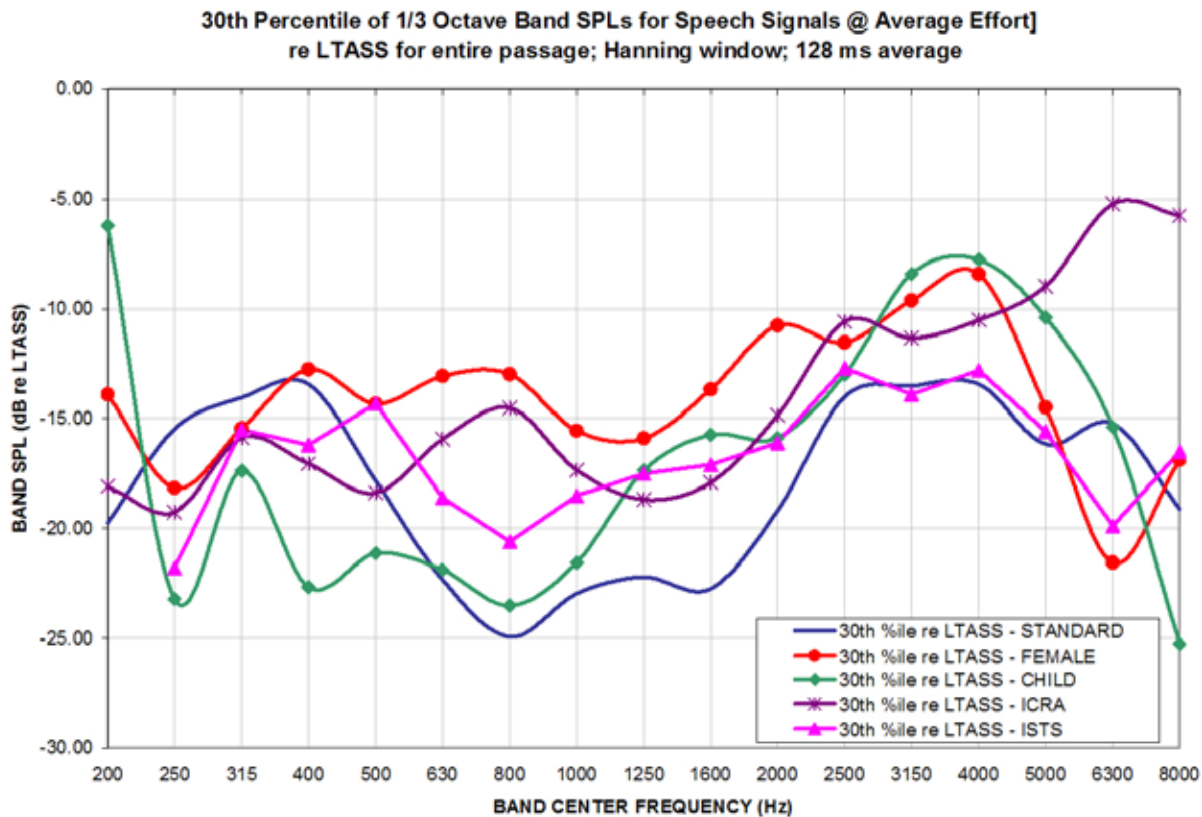


Figure 14: Level of 30th percentile for Speechmap speech signals at average vocal effort

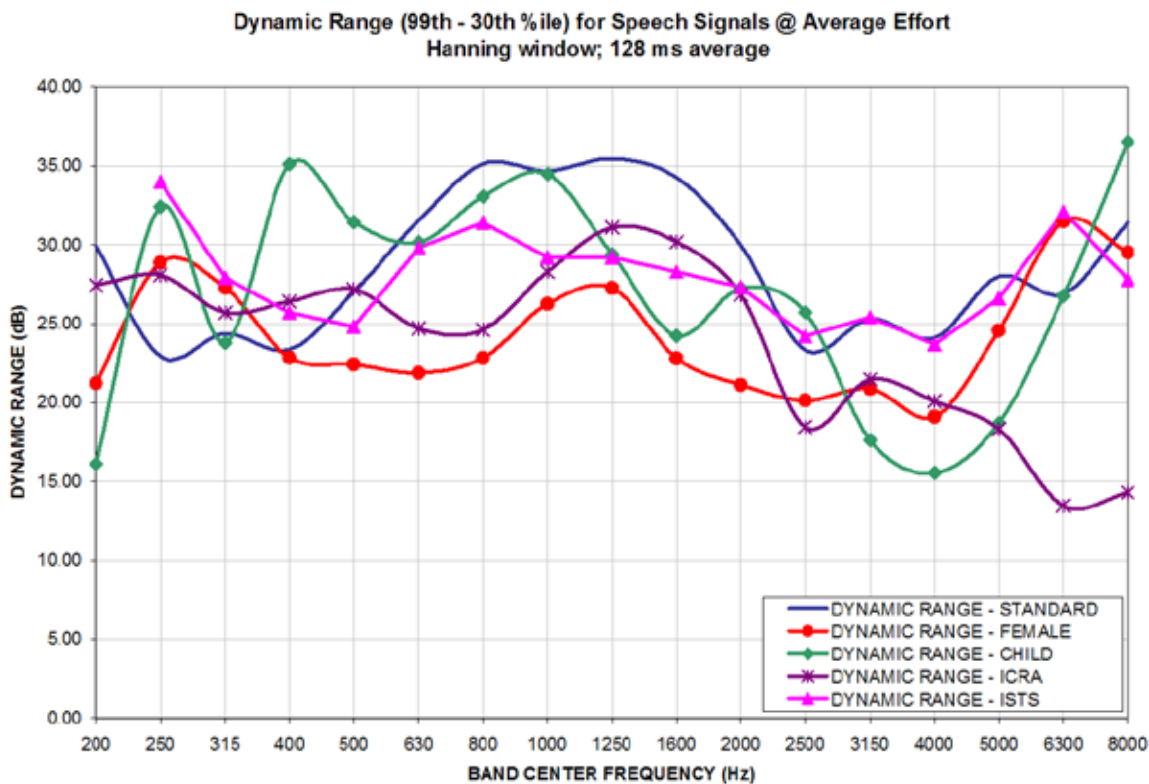


Figure 15: Dynamic range of Verifit speech signals at average vocal effort