- [ ] 2700 Series
- [x] APx555
- [x] APx585 Series
- [x] APx525 Series
- [x] APx515

Measuring Speech Transmission Index with APx audio analyzers

by Joe Begin

About this Technote

In this Technote, we discuss the Speech Transmission Index (STI) and how to measure it with an APx500 Series audio analyzer. Our focus (beginning on page 2) is on the widely-used, simplified STI method called STIPA.

About STI

Speech Transmission Index is an objective measure used to predict the intelligibility of speech transmitted from a talker to a listener through a "transmission channel." A transmission channel, in this respect, can be as simple as the unamplified acoustic path between a speaker and a listener in the same room. Or, it could be more complex, incorporating electronic and electro-acoustic components such as amplifiers, compressors, microphones, loudspeakers, codecs,* radio links, etc. In the STI method, a special test signal is applied to the transmission channel and the received signal is analyzed to derive a single number rating between 0 and 1, representing the potential for speech intelligibility through the channel.

The STI method was first introduced in the 1970s, and has been the subject of ongoing development and improvement since that time. Although it has some limitations, the STI method has proved useful in many situations and has gained international acceptance.

IEC 60268-16: the STI Standard

The definitive international standard for STI is *IEC 60268-16—Sound System Equipment Part 16: Objective Rating of Speech Intelligibility by Speech Transmission Index*. This standard was first published in 1988, and as of 2011 is in its 4th revision. The latest version contains a wealth of information on STI including theory, the history of STI development, variations of the STI method, their application and limitations, and recommended procedures for measurement and post-processing of data.

Background

Conceptually, STI is based on the empirical finding that the normal fluctuations in speech signals carry the most relevant information related to speech intelligibility. These fluctuations (or *modulations*) result from the acoustic separation of sentences, words, and phonemes (the fundamental elements of speech). For clear speech, modulation rates extend from 0.5 Hz to 16 Hz, with maximum modulation at about 3 Hz.

Fluctuations in speech can be quantified in terms of modulation depth and modulation rate (frequency), which together form a modulation spectrum. STI uses the concept of the modulation transfer function (MTF), which is the ratio of modulation depth of the received signal to the modulation depth of the transmitted signal as a function of frequency. Any reduction of the MTF by the transmission channel is considered to result in some deterioration of speech intelligibility. Distortions in the transmission channel which can reduce the modulation depth (and hence intelligibility) include noise, reverberation, echoes, nonlinear distortion, spectral distortion, and digital codecs.

The spectral content of speech itself is also important. The long-term average speech spectrum includes the octave bands from 125 Hz through 8 kHz, but these frequencies do not contribute equally to intelligibility. Although low speech frequencies (which make up vowel sounds) contribute most to the power of a speech signal, it is the higher speech frequencies (consonants) which contribute the most to speech intelligibility. This is reflected in the weighting factors applied when calculating the STI (Figure 1).

* Encoder/decoder combination.

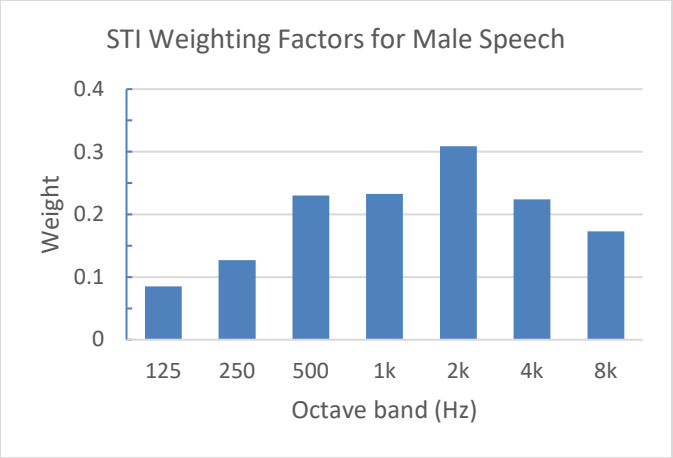


Figure 1. Weighting factors used in STI calculation. These are indicative of the relative contribution to intelligibility of the octave bands containing speech.

In general, an STI signal is formed from seven octave-band noise signals, with overall spectral shape matching the long-term average spectrum of speech. Each of these noise carriers is modulated with one or more of 14 low frequency sine waves at modulation rates from 0.63 Hz to 12.5 Hz, spaced at ⅓-octave intervals. The STI signal is passed through the transmission channel with the output signal being captured at a listener position. From this output signal, the modulation transfer ratio (the reduction in modulation depth) is determined and used to calculate a single-number STI value.

STI Methods

Full STI

The full STI method uses all combinations of seven noise carriers and 14 modulation rates (a total of 98 values) to determine the MTF of a transmission channel. Results are averaged for each of the seven octave bands and weighted according to the band's relative contribution to speech intelligibility. These weighted values are then summed to determine the overall single-number (0 to 1) STI value for the transmission channel.

The full STI method requires 98 test signals to be applied sequentially, with each measurement taking about 10 seconds. The overall measurement time is approximately 15 minutes. Full STI is considered to be the most accurate of the STI methods, but it is rarely used today because simplified methods which were subsequently developed are faster and more practical. Of these simplified methods, STI for Public Address systems (STIPA) appears to be the most popular, and is the focus of this Technote. Other simplified methods include RASTI and STITEL. RASTI is now considered to be obsolete. STITEL is currently in use, but great care is required when using it.[1]

Indirect STI Methods

The full STI method and its simpler variants, like STIPA and STITEL, are referred to as *direct* STI methods because they use modulated speech-shaped noise signals to directly derive the MTF, which is the basis of the STI. The MTF can also be derived from the measured impulse response of a transmission channel using the Schroeder technique referenced by Rife.[2] Methods which derive the MTF from the impulse response are referred to as *indirect* methods. These apply only to linear time-invariant systems and, according to IEC 60268-16,[1] demand considerable experience in their use because such measurement systems typically require correct adjustment of many measurement parameters that could affect the final result.

STIPA

A simplification of full STI, STIPA is a method in which each of the seven octave bands contains a predefined set of two modulation frequencies. These 14 combinations are generated simultaneously in one signal and are processed in parallel, resulting in a significantly reduced measurement time of 10 to 20 seconds. In spite of its much shorter measurement time, the STIPA method has been found to compare very favorably with full STI in a large variety of test conditions which might be encountered in PA systems.[3] The STIPA method has only been validated for the male speech spectrum.

IEC 60268-16 provides comprehensive guidance on the types of distortion each STI method is appropriate for, and the applicability of the various methods for different uses.

STI Applications

Potential applications for STI include:

- Public Address (PA) systems
- Voice Announce (VA) and emergency sound systems
- Communication channels & systems
  - Intercoms
  - Wireless systems
- Rooms and auditoria

- Direct (unamplified) communication in rooms, including vehicles
- Assistive hearing systems

STI should not be used for transmission channels containing vocoders (codecs which operate specifically on speech elements), but it may be used for digital codecs that operate on the entire signal. It also cannot be used on systems with aggressive noise-suppression algorithms, because the STI signal itself is modulated noise that is likely to be suppressed by the algorithm. Recent research attempting to measure STI with real speech signals shows promise, but is not yet commercially available.[4]

Industry Standards

Some examples of industry standards where STI is used or referenced are:

- *NFPA 72: The National Fire Alarm Signaling Code, Annex D, speech intelligibility* (informative)
- *NFPA 1982: Standard on Open-Circuit Self-Contained Breathing Apparatus (SCBA) for Emergency Services* (voice communication system performance requirements)
- *EUROCAE ED-112A: Minimum Operational Performance Specification for Crash-Protected Airborne Recorder Systems*
- *UK-CAA Specification No. 15: Aircraft Public Address Systems*
- *RTCA DO-214A: Aircraft Audio Systems*

STI versus PESQ or POLQA

On the surface, it would appear that STI is similar to perceptual audio measurements like PESQ and POLQA. However, there are some important differences: STI is a metric for speech *intelligibility*, whereas PESQ and its newer cousin POLQA are measures of speech *quality*.

Speech intelligibility is defined as the degree to which a normal person can understand speech, whereas speech quality is defined as the difference between idealization (the theoretical capabilities of a communication channel) and realization (the performance of the channel in practical use). The two are related, but they are not the same. An example of a transmission channel with high quality but low intelligibility is a system with high bandwidth but a tendency to drop entire words, as can occur in some VoIP (voice over internet protocol) systems. The extended bandwidth provides higher quality, enabling the speech to sound much more natural, but when words are dropped, intelligibility is poor. In contrast, a lower bandwidth system without drop-outs, such as a POTS (plain old telephone service) telephone line, will have lower quality but higher intelligibility.

There is not a lot of information available comparing STI to PESQ or POLQA, but it would seem that the latter two measurements are more applicable to quiet listening conditions, typical of telephone conversations, for which these measurements were designed. Given its use in public address, aircraft and emergency communication systems, STI seems more applicable to high-noise environments, where intelligibility is paramount and speech quality is of secondary concern—provided that vocoders or aggressive noise suppression algorithms are not in use.

STI Measurement in APx500

The STI functionality for APx500 Series audio analyzers is distributed as a measurement "plug-in," meaning it is not built in to the software. To add the capability to APx, users must first download* and run the plug-in installer. Once it has been installed, two new measurements appear in the Add Measurement dialog in Sequence mode (Figure 2): Speech Level (IEC 60268-16J) and Speech Transmission Index (STIPA).

> Note: The STI plug-in is a software option and requires a software key for measurement functionality; the plug-in can be demonstrated without a software key in APx Demo Mode. The APx STI plug-in requires v4.2.1 or later of APx500 audio measurement software.

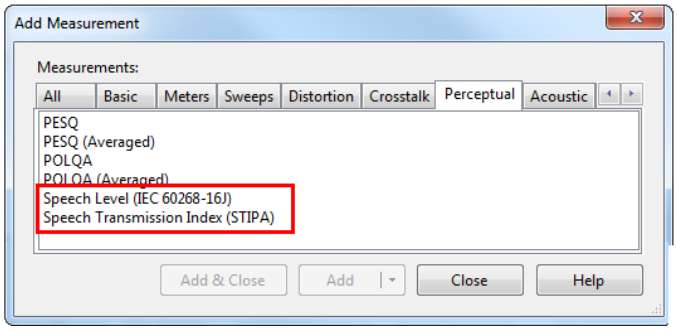


Figure 2. Add Measurement dialog (the two STI measurements may also be found in the "All" tab).

* Download plug-in at: http://www.ap.com/

As its name implies, the Speech Transmission Index (STIPA) measurement is used to conduct an STI measurement using the STIPA method. The measurement controls are shown in Figure 3.

The STIPA Signal

The STIPA measurement requires a special signal containing shaped noise filtered in ½-octave bands and modulated with two modulation frequencies per band. The signal has a unique FFT spectrum, as seen in the APx signal monitor view in the lower left corner of Figure 3. When an STI measurement is added to a signal path in APx, several versions of the STIPA signal are automatically loaded into the project file as .wav files. These files are named in the form "AP_STIPA_ss_bb_vx.y.wav" where "ss" represents the .wav file sample rate in kilohertz, "bb" represents the bit depth (typically 16 bits) and "x.y" represents the version number. STIPA signals at five sample rates are included: 48 kHz, 44.1 kHz, 24 kHz, 16 kHz and 8 kHz. The default is set to the waveform with 24 kHz sample rate, because this sample rate is just sufficient to encompass the upper frequency range of the STIPA signal.

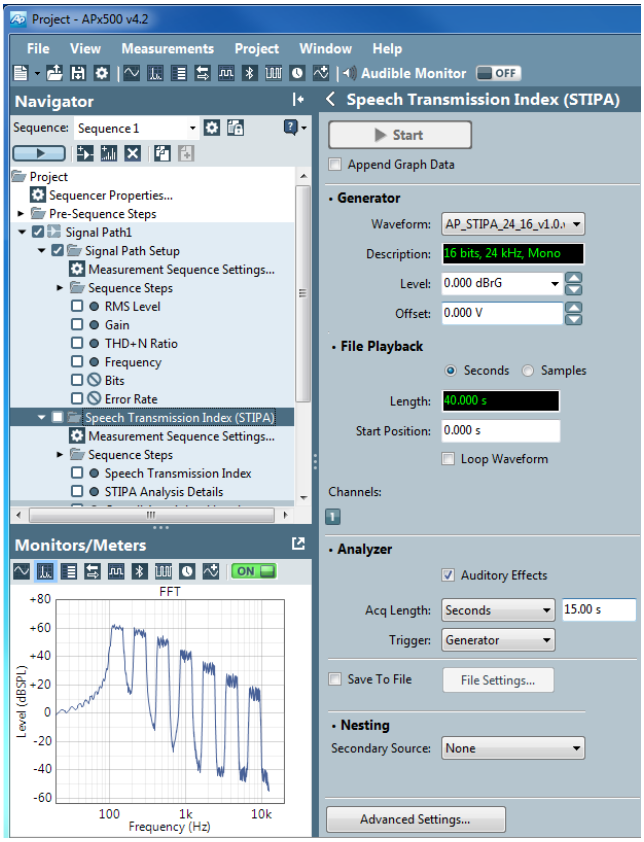


Figure 3. The Speech Transmission Index (STIPA) measurement in APx.

The different sample rates are included mainly to accommodate generating the STIPA signal over *Bluetooth*™. When Bluetooth is selected as the output "connector" in APx, the sample rate of any arbitrary waveforms generated must match the system output sample rate, which for Bluetooth is typically 44.1 kHz or 48 kHz in the A2DP profile and 8 kHz or 16 kHz in the HFP profile. See page 8.

When the APx output is set to Analog or any other Digital interface, the system will automatically use the sample rate of the selected arbitrary waveform file. In this case, a waveform with a sample rate of 24 kHz or greater should be selected, unless a lower sample rate is specifically required.

Setting the Signal Level

Once the signal path is configured using Signal Path Setup, set the generator level for the STI measurement as required. Note that in APx, the generator level control is calibrated for sine waves, and the AP STIPA signal is a type of noise waveform with an rms level approximately 15 dB below full scale, to accommodate the crest factor of the signal. As a result, some experimentation may be required to find an appropriate value for the generator level control.

STI Measurement Analyzer Controls

The Acquisition Length for the measurement is set to 15 seconds by default. The user can enter any value into this field, but IEC 60268-16 requires a measurement time of between 10 and 20 seconds*.

The Trigger control is set to Generator by default, which is appropriate when a waveform file like the STIPA signal is being generated. This will ensure that acquisition starts concurrent with the waveform signal generation.

The Auditory Effects checkbox is used to enable or disable the inclusion of auditory effects in the STI calculation. These effects include level-dependent auditory masking and the absolute speech reception threshold, as defined in IEC 60268-16. These effects can only be included when the APx system is in Acoustic input mode (i.e., when the system is measuring sound pressure level).

* A default value of 15 seconds was selected as a compromise due to the ambiguity of IEC 60268-16 in this regard: in the main text, the standard states that the STIPA measurement time is 10 to 15 seconds while Annex B (STIPA) indicates a range of 15 to 20 seconds is required.

Another default setting in the STI measurement not immediately obvious to users is the input sample rate setting of 24 kHz (11 kHz bandwidth). This sample rate was selected as the default because it encompasses the upper limit of the STIPA signal (8 kHz + ¼ octave), and there is no need to include higher frequencies in the analysis. To use a different rate, choose a different bandwidth in the Advanced Settings dialog.

The STI measurement controls also include a "Save to File" checkbox which can be used to save the acquired waveform to a file for subsequent analysis.

Note: The Nesting control visible below the Analyzer controls should be ignored for the STI measurement.

Signal Path Setup

Before running the STI measurement, Signal Path Setup should be used to configure the analyzer outputs and inputs.

For Output Configuration, this includes the choice of Connector, Acoustic mode and EQ (equalization). When testing systems whose input is acoustic (e.g., a microphone), the analyzer's output EQ feature can be used to properly equalize the loudspeaker used to apply the stimulus.

For Input Configuration, the choices include Connector, Bandwidth, Acoustic mode, Filters, and EQ. When testing systems whose output is acoustic (e.g., loudspeakers or headphones), the analyzer's Acoustic input mode can be used to display results in dBSPL directly, once the microphone(s) have been calibrated. Input EQ is also available for microphones with a non-flat frequency response. * A special case of this is the drum reference point (DRP) to free-field correction, which is required if ear simulators (e.g., IEC 60711 couplers) are used. An example of such a case would be testing headphones or earphones using a head and torso simulator (HATS).

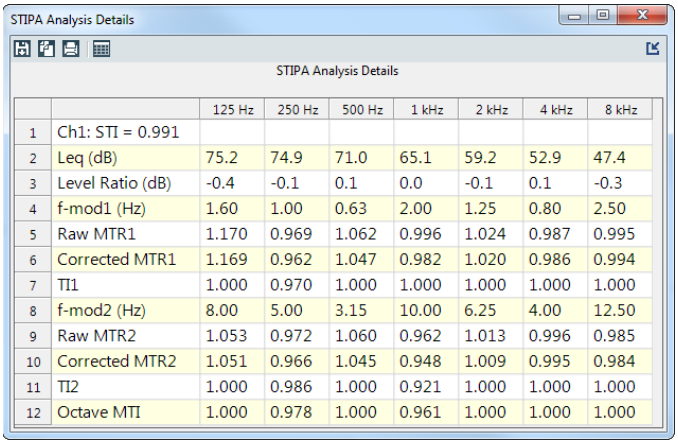
Running the STI Measurement

The STI measurement can be run by simply clicking the Start button. This will generate the selected signal through the device under test (DUT) or transmission channel, and acquire the output signal for the specified acquisition length in seconds. Once the acquisition is complete, the system will conduct the STI analysis and display the results.

Like any measurement in APx, the STI measurement can also be run from the Sequencer. When the Sequencer is used, results are checked against limits (if applied) and the data for any checked results are included in the Sequence Report.

STI Measurement Results

The STI measurement has two meter (single value) results: Speech Transmission Index and Overall A-weighted Level (provided for convenience). A third result is a table called STIPA Analysis Details (Figure 4). In addition to the overall STI value for each measurement channel, this table includes the detailed intermediate results described in Table 1.



		125 Hz	250 Hz	500 Hz	1 kHz	2 kHz	4 kHz	8 kHz
1	Ch1: STI = 0.991							
2	Leq (dB)	75.2	74.9	71.0	65.1	59.2	52.9	47.4
3	Level Ratio (dB)	-0.4	-0.1	0.1	0.0	-0.1	0.1	-0.3
4	f-mod1 (Hz)	1.60	1.00	0.63	2.00	1.25	0.80	2.50
5	Raw MTR1	1.170	0.969	1.062	0.996	1.024	0.987	0.995
6	Corrected MTR1	1.169	0.962	1.047	0.982	1.020	0.986	0.994
7	TI1	1.000	0.970	1.000	1.000	1.000	1.000	1.000
8	f-mod2 (Hz)	8.00	5.00	3.15	10.00	6.25	4.00	12.50
9	Raw MTR2	1.053	0.972	1.060	0.962	1.013	0.996	0.985
10	Corrected MTR2	1.051	0.966	1.045	0.948	1.009	0.995	0.984
11	TI2	1.000	0.986	1.000	0.921	1.000	1.000	1.000
12	Octave MTI	1.000	0.978	1.000	0.961	1.000	1.000	1.000

Figure 4. The STIPA Analysis Details result.

Table 1. Description of Results in Figure 4

Item	Description
Leq (dB)	The overall rms level for each octave band (referred to as the equivalent sound level for acoustic inputs)
Level Ratio (dB)	Indicates the spectral shape of the received signal relative to the source signal, normalized to the 1 kHz octave band.
f-mod* (Hz)	The modulation frequency.
Raw MTR*	The calculated raw modulation transfer ratio.
Corrected MTR*	The MTR after correction for auditory masking and reception threshold.
TI*	The transmission index.
Octave MTI	The average of TI1 & TI2

* *These items have two values for each octave band*

* The STI method is based on measurements made in a free field. A free-field microphone should be used, or alternatively, a free-field correction should be applied.

Speech Level Measurement

The STI plug-in also adds a measurement named Speech Level (IEC 60268-16J) which determines the "speech level" as defined in Annex J of that standard. It applies A-weighting to the speech signal and removes the silent parts between words before calculating the level. The resulting speech level is a better estimate of the true signal level that can be used to select an appropriate level for the STIPA signal. The intent is similar to that of the Active Speech Level concept used in Recommendation ITU-T P.56.[5]

The Speech Level measurement has four results: Acquired Waveform, Frame Levels (A-weighted), Speech Level (A-weighted) and RMS Level (A-weighted). After A-weighting the signal, the measurement divides the signal into frames of 10 ms to 20 ms in length and calculates the rms level of each frame. The Frame Levels result is a plot of these levels versus time.

The Speech Level measurement loads a speech waveform named "Default 48 kHz.wav", but any waveform can be loaded using the Browse for file selection of the Waveform control. This measurement also allows the acquired waveform to be captured to a WAV file.

Open-Loop Measurements

APx software also supports open-loop measurements, in which the signal originates from a different source than the audio analyzer (e.g., a CD player or portable media player). In this mode, the Output Connector in Signal Path Setup is typically set to **None (External)**, which eliminates the generator controls from each measurement.

For open-loop STIPA testing, the AP STIPA signals can be exported as WAV files from the APx project file using the File > Manage Attached Project Items menu choice. Once exported, these WAV files are available for generation from the external source.

> Note: The APx STIPA measurement can also be used with third party STIPA signals, provided those signals conform to IEC 60268-16 (2011).

File Analysis

Another very useful feature within the APx software is called File Analysis. This mode of operation is entered by setting the Input Connector in Signal Path Setup to either File (Digital Units) or File (Analog Units). Once set, any measurement can be used to analyze the contents of a series of WAV files, just like an acquired input signal (Figure 5). The Analog Units choice is intended for WAV files recorded with APx, which are then scaled back to their original voltage levels.

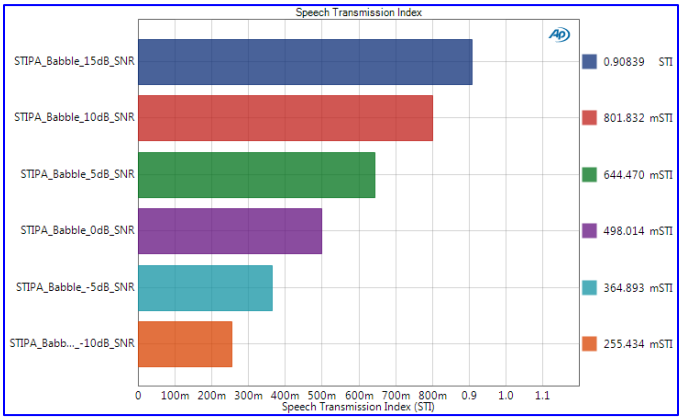


Figure 5. STI analysis of WAV files of the STIPA signal mixed with noise at various Signal-to-Noise ratios.

STI Measurement Examples

PA System with Microphone Input

When testing the microphone input of a PA system, a special loudspeaker (e.g., a mouth simulator) that emulates a person speaking is needed to apply the STIPA signal. The best choice in this regard is a HATS with an artificial mouth.

No loudspeaker has a flat frequency response, so equalization is required to ensure that the signal has the correct spectral shape. IEC 60268-16 specifies that the mouth simulator be EQ'd within ±1 dB of the specified STIPA signal shape within individual octave bands from 125 Hz to 8 kHz, when measured in a free field. For this application, because the response within octave bands is required, it is easiest to use a sparse EQ curve with frequencies at the standard octave band center frequencies. The EQ values can then easily be adjusted by trial and error, until the required spectral shape is achieved (Figure 6).

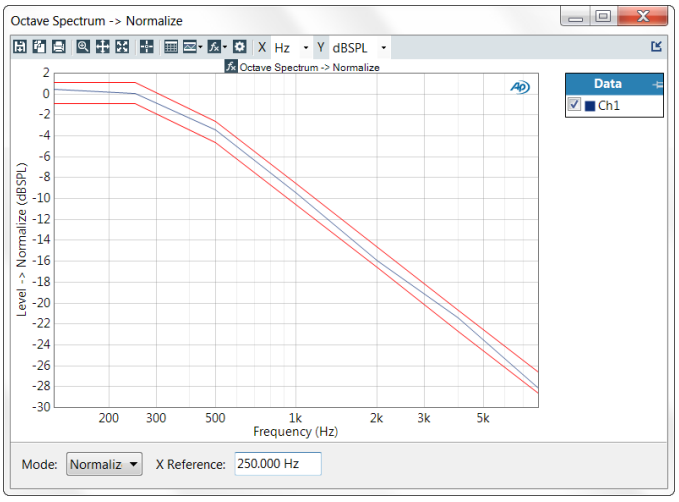


Figure 6. Equalizing the STIPA signal within octave bands.

The level of the test signal should be adjusted to achieve an appropriate level as measured at the PA system microphone. One way to do this is to generate speech signals at a typical operational level (e.g. for comfortable listening), and then use the Speech Level measurement to measure the resulting A-weighted speech level. The STIPA signal is then adjusted to achieve the same overall A-weighted level. Alternatively, as noted in IEC 60268-16, a default equivalent level of 60 dBA at 1 m in front of the mouth simulator can be used.

> Note: Depending on the loudspeaker used, an aggressive EQ curve may be required. This, combined with the −15 dBFS rms level of the STIPA signal and its broadband nature, may require a higher level of amplifier gain than is typical for sine and speech signals. However, care should be exercised to avoid overloading the mouth simulator.

For PA system measurements, the inputs to the analyzer will typically be measurement microphones, in which case the Acoustic input mode in APx can be selected with the microphones calibrated to read in dBSPL. For acoustic inputs, Auditory Effects should be enabled using the check box in the APx STI measurement control panel. According to Annex G of IEC 60268-16 (STI qualification bands), the minimum target STI values for “good quality” and “high quality” PA systems are 0.54 and 0.58, respectively.

NFPA 1981

This standard[6] covers self-contained breathing apparatus (SCBA) for emergency services. It has performance requirements for both non-electronic and supplementary (amplified) voice communications. The test methods for both non-electronic and supplementary systems are quite similar. Both methods require a hemi-anechoic chamber and a HATS with mouth simulator (Figure 7).



Figure 7. SCBA mounted on a G.R.A.S. KEMAR manikin in an anechoic chamber. [Photo courtesy of Scott Safety, used by permission; for illustration only, not to scale.]

These NFPA test methods require measuring STI at a test microphone 1.5 m in front of the artificial mouth while simultaneously generating pink noise from a separate speaker located below the test microphone (Figure 8).

NFPA 1981 requires that the mouth simulator be equalized to a specific ⅓-octave spectral shape (Figure 9) at the mouth reference point (MRP) of the mouth simulator. In this case, the output EQ curve in the APx software can be specified at standard ⅓-octave center frequencies, for convenience when adjusting the EQ by trial and error.

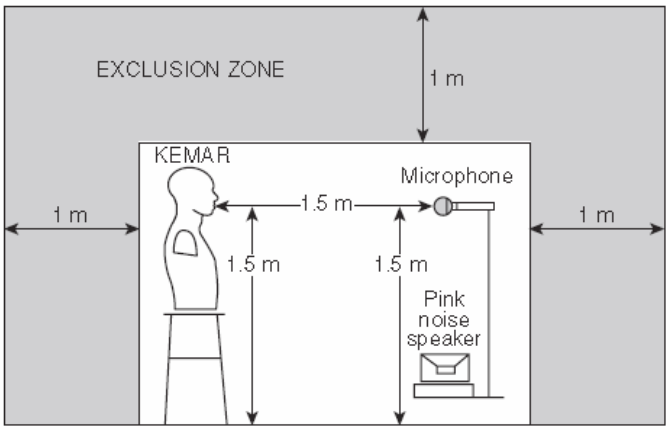


Figure 8. Test setup within the anechoic chamber [from Figure 8.10.4.9(b) of reference 6].

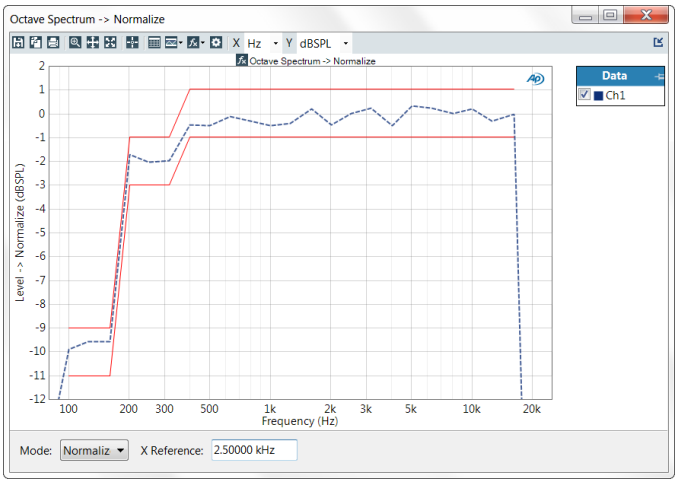


Figure 9. Spectrum shape at mouth simulator MRP.

> Note: NFPA 1981 suggests using a STIPA signal generator with an equalizer to drive the mouth simulator and a separate pink noise generator with an equalizer to drive the pink noise speaker. With APx, both signals can be generated simultaneously by the audio analyzer by taking advantage of its ability to generate stereo waveforms with different levels on each channel. To use this "trick," you need to record each signal as a .wav file after the necessary EQ is applied and then combine them into a stereo .wav file using 3rd party waveform editing software.

NFPA 1981 requires a minimum STI of 0.55 for non-electronic systems and 0.60 for supplementary (amplified) systems.

Bluetooth Headset Receiving over HFP

For this example, we tested a Bluetooth headset with optional noise cancellation in receive mode in the Hands Free Profile (HFP). HFP currently has two modes of operation: Narrowband voice, with a sample rate of 8 kHz, typically uses the CVSD codec, and wideband voice, with a sample rate of 16 kHz, typically uses the mSBC codec.

The reduced bandwidth of the narrowband voice mode (about 3.7 kHz) has a significant impact on the potential STI score, because the entire 8 kHz octave band and a portion of the 4 kHz octave band are eliminated from the analysis (Figure 10). Because of this, the highest STI value that can be measured with the narrowband signal under perfect conditions (with the signal looped back) is about 0.92. The wideband voice mode fares much better in this regard, with a near-perfect score in loopback mode, even though almost half of the 8 kHz octave band is eliminated from the signal.

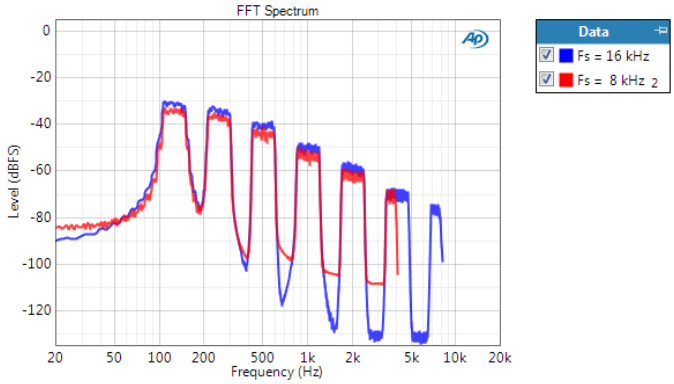


Figure 10. FFT Spectrum of STIPA signals at 16 and 8 kHz sample rates.

To test the headset in receive mode, a HATS (head and torso simulator) with ear simulators is required. The Acoustic input mode of APx is used and the ear simulators are calibrated so that the system measures in dBSPL directly. In addition, when using the STI method, an input EQ curve should be used to correct the measured spectrum from the drum reference point to the free field.

An APx audio analyzer with a Bluetooth module is required for this test case. To generate the signal, you must first set the Output Connector to Bluetooth in Signal Path Setup and select the profile set A2DP Source, HFP Audio Gateway, AVRCP Target. This tells the APx system to simulate an audio gateway (e.g., a Bluetooth-enabled phone). Then you can put the device in pairing mode, scan for it, and Pair and Connect the HFP profile. Alternatively, you can pair with the device without scanning by entering its Bluetooth address. Finally, to open a communication channel to the headset from the Actions dialog, you can either open the SCO (synchronous connection-oriented link), or ring the device and use its answer button to simulate a call.

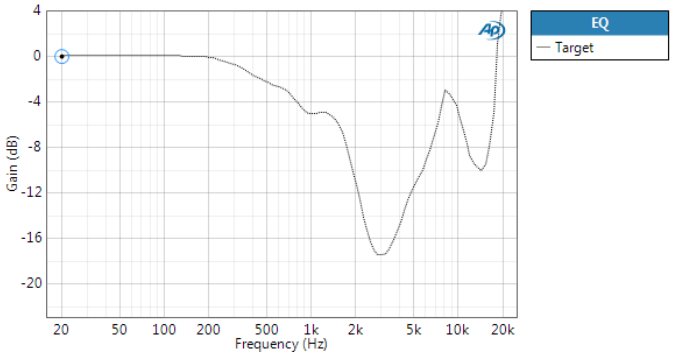


Figure 11. DRP to free field correction implemented as an Input EQ curve in APx.

As mentioned above, when generating the STIPA signal over Bluetooth, a version of the waveform matching the system

output sample rate must be selected. Our example Bluetooth headphones used narrowband voice, which requires the STIPA waveform at 8 kHz sample rate.

In the absence of a specification, you can set the signal level on the headphones to achieve a comfortable listening level for yourself or another person before putting them on the HATS to conduct the test. Proceed as follows: While wearing the headphones, Use the Speech Level measurement to generate one or more speech signals and adjust the headset volume controls and/or the APx generator level for comfortable listening. Then put the headphones on the HATS and measure the A-weighted Speech Level without adjusting the signal level. Finally, run the STI measurement a few times and adjust the generator level until approximately the same A-weighted level as the speech signals is measured.

The Bluetooth headphones we tested achieved an STI value of 0.75 with no ambient noise in the room. When the ambient noise level was increased to 75 dBA by generating pink noise from a loudspeaker, the measured STI values were 0.61 with the headset's noise cancellation feature disabled and 0.71 with it enabled.

References

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