Audio Engineering Society

Convention Express Paper 82

Presented at the 154th Convention

2023 May 13-15, Espoo, Helsinki, Finland

New method for the computation of acoustic parameters according to the updated Italian Legislation

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ABSTRACT

Standard acoustic parameters such as reverberation time T_{20} and clarity C_{50} are measured or calculated according to ISO 3382-1. The additional parameter STI (Speech Transmission Index) is measured or calculated according to the IEC 601268-16 standard, recently upgraded in 2020. In Italy, it is a legal requirement to verify the values of these acoustic parameters for new public buildings, both in the design phase (calculated values) and in the official assessment of the building performance (measured values). The relevant legislation was drafted in 1997, updated in 2017 with the introduction of the minimum environmental criteria (Decree of the Ministry of the Environment of 11/10/2017)), and recently further updated with the Decree by Ministry of Ecological Transition No. 256 of 23/06/2022. This decree became legally effective in December 2022 and is still very innovative at the time of writing. It requires that the values of three "advanced" acoustic parameters (T_{20}, C_{50} and STI) be evaluated in all new public buildings, with particularly severe prescriptions in the values to be achieved in special categories of buildings, such as schools.

These advanced acoustic parameters are additional to the standard ones, such as the sound reduction index of partitions and facades, the impact sound level and the noise level generated by HVAC systems, which are mandatory in Italy for all types of buildings since the introduction of DPCM 5/12/1997. This paper deals with these three advanced acoustic parameters, which have only recently become mandatory, and describes the update of the Ramsete and Aurora software packages, widely used in Italy to perform predictive calculations and experimental verification of these "room acoustic" parameters. Recent updates to these software tools make them compatible with the newer version of the standards, and hence with the new legislation. In particular, these developments now include a new method for calculating the room impulse response, and hence the STI value, taking into account the different orientations of the speaker, synthesizing the corresponding voice directivity balloon. This new STI(θ) measurement allows the evaluation of the speech intelligibility index for the case when the speaker is not facing the listener. This result can be obtained both experimentally and in computer simulations by evaluating the MIMO Room Impulse Response, and then synthesizing a virtual source with the standardized human voice directivity and proper aiming.

1 Introduction

The acoustic comfort of a space is defined as a satisfactory sound perception in a given environment, which must be considered under qualitative and quantitative conditions based on the balance between sound absorption and reverberation. These two characteristics are directly related to the type of materials used for room finishing and the volume size [1].

Speech intelligibility is an important requirement, especially in educational buildings and auditoriums, since the conditions for concentration and attention are highly dependent on how much words can be understood [2].

Numerous studies have shown how unfavourable listening conditions can affect speech comprehension and lead to poorer academic performance, which can be even worse if the listener or speaker are not native language [3].

Speech intelligibility is also determined by background noise levels, which are as important as reverberation; a comfortable space for speech occurs when the signal-to-noise ratio (SNR) is consistent [4]. Therefore, both mechanical ventilation noise and break-in noise from outdoor environment should be under control and below the limits set by standard criteria.

The acoustic comfort of a room for speech should also be suitable for speakers who are sometimes in a state of excessive vocal effort, when the acoustic treatment of a room is not satisfactory [5].

There are several solutions to control the acoustics of a room: absorbing panels are mainly used for mid-high frequencies due to their wider absorption [6], while resonant structures are used for low frequencies to mitigate booming effects [7]. The innovation of the technological material leads to the development of micro-perforated panels, which can also be installed with some cavity instead of being directly attached to the back wall [8].

This paper deals with the description of the acoustic parameters mainly used for speech in different conditions. The acoustic comfort of rooms is determined by a variety of criteria specified in regulations and guidelines, depending on the function of the room, as being categorised for each type of building, and the volume of the room.

In ISO 3382-1:2009 [9] techniques for measuring acoustic parameters from a Room Impulse Response (RIR) are described. The most commonly used techniques are the measurement of impulse response using sine sweep method [10] or the use of an impulsive source [11]. Both techniques require post-processing analysis to determine the acoustic parameters of the room, namely reverberation time (T_{20}), speech clarity index (C_{50}). Another relevant parameter is Speech Transmission Index (STI), which is defined in the IEC 60268-16:2020 standard [18]. These three acoustic descriptors have been selected by the Italian legislator to ensure adequate acoustic quality in public buildings, and especially in schools.

2 Acoustic parameters related to speech

There are some acoustic parameters that describe the quality of a sound field related to speech. The quality of a room depends on the balance between early and late reflections arriving at the receiver [12]. This concept led to the determination of T_{20}, C_{50}, and STI. Reverberation time is expressed in seconds and refers to the time it takes for sound energy to decay by one millionth compared to the initial level at the time of a stationary sound source, which corresponds to a decay of 60 dB. Reverberation time was the first physical parameter developed to describe the acoustic conditions of an enclosed space and can be predicted with a reasonable accuracy using the Sabine's formula, as shown in equation (1).

$$T_{60} = 0.16 \frac{V}{\sum_i (\alpha_i S_i)} \tag{1}$$

where *V* is the volume of a space, α is the absorption coefficient assigned to the materials, and *S* is the surface area related to each material in a room.

Since decay over 60 dB is difficult to be achieved in many environments, T_{20} is used as an alternative, extrapolated by the sound energy decay by 20 dB, from -5 dB to -25 dB below the steady-state level. T_{20} is still expressed as a decay of 60 dB and is usually expressed in octaves or one-third octave bands over the entire bandwidth. In Italy, the reference standard for describing acoustic properties in schools is UNI ISO 11532-2 [13].

Speech clarity is evaluated using the C_{50} parameter. Equation (2) shows the definition of C50, according to ISO 3382-1:2009.

$$C_{50} = 10 \cdot log\left(\frac{\int_0^{50ms} p^2(t)dt}{\int_{50ms}^{\infty} p^2(t)dt}\right) \tag{2}$$

where *p(t)* is the pressure impulse response measured by an omnidirectional microphone. For measurement of clarity C50, the sound source is also usually assumed to be omnidirectional, and typically a dodecahedron loudspeaker is used. C_{50} represents the ratio of reflections arriving within 50 ms to reflections arriving after 50 ms. The optimal values for C_{50} should be between -2 dB and +2 dB [14]; at values above this range limit, the sound in the room is perceived as dry, while values below this range limit indicate an excess of reverberation.

The acoustic parameter used to determine speech intelligibility is STI, calculated from the modulation transfer function (MTF) [16], defined as the ratio between the amplitude of the modulation at the receiver and the corresponding modulation of the sound signal generated by the source, which is a filtered pink noise in the seven octave bands between 125 Hz and 8 kHz. This particular signal is used to mimic the human voice in a real environment, which may be affected by background noise and reverberation. By varying the modulation frequency between 0.63 Hz and 12.5 Hz, a total of 98 MTF values are obtained, which are averaged to obtain a single representative STI value.

The IEC 60268-16:2020 standard [18] describes all the instructions for measuring the STI parameter, from the calibration of the talk-box with a determined spectrum to the signal processing required to output the MTF and STI results.

The calculation of MTF and STI is possible from the measured impulse response (IR), filtered in octave bands, as studied by Schröder in 1981 [17], as shown in equation (4).

$$MTF = \frac{\int_0^\infty h_f^2(\tau) \cdot e^{-j2\pi F\tau} d\tau}{\int_0^\infty h_f^2(\tau) d\tau} \tag{3}$$

The result obtained with equation (3) does not take into account the background noise in a room, but only the reverberation effect. For this reason, a factor must be added to equation (3), taking into account the effect of the SNR, as shown in equation (4).

$$MTF = \frac{\int_0^\infty h_f^2(\tau) \cdot e^{-j2\pi F\tau} d\tau}{\int_0^\infty h_f^2(\tau) d\tau} \cdot \frac{1}{1 + 10^{\left(\frac{L_{noise} - L_{signal}}{10}\right)}} \tag{4}$$

This equation shows that high SNR and a low reverberation time are required for good speech intelligibility.

The categories for STI quality are summarised in IEC 60268-16 [18] and are listed in Table 1.

STI Range	Speech Quality
0 < STI < 0.3	Bad
0.3 < STI < 0.45	Poor
0.45 < STI < 0.60	Fair
0.60 < STI < 0.75	Good
0.75 < STI < 1.0	Excellent

Table 1. Speech Quality vs STI according to IEC 60268-16.

3 Overview on the Italian legislation

The Decree of June 23, 2022, also known as Decree MITE No. 256, revises the minimum environmental criteria (CAM) related to the acoustic performance of public spaces, focusing on schools, hospitals, and healthcare buildings. This decree imposes to comply with the values of the acoustic parameters and their optimal range as described in UNI 11367:2023 [15], for all public buildings except schools; for them, the UNI 11532-2:2020 standard must be applied, which generally prescribes more restrictive values for the acoustic parameters.

According to UNI 11367:2023, the acoustic parameters that should be considered for spaces where speech is the predominant function are T20, C50 and STI. Table 2 summarizes the criteria set by UNI 11367.

	T_{20} (s)	C_{50} (dB)	STI (dB)
Places for speech	See eq. (5)	≥ 0	≥ 0.6
Sport facilities	See eq. (6)	≥ -2	≥ 0.5

Table 2. Recommended values for C_{50} and STI parameters, based on UNI 11367:2023.

The optimal values of reverberation time, calculated for the middle frequency bands between 500 Hz and 1 kHz are given in equations (5) and (6).

$$T_{ott} = 0.32 \cdot \log(V) + 0.03 \tag{5}$$

$$T_{ott} = 1.27 \cdot \log(V) - 2.49 \tag{6}$$

where V is the volume of the room.

Another code that applies specifically to schools and similar buildings is UNI 11532-2:2020, which classifies all building types according to their space function. Each category has different STI criteria related to the size of the space. In general, the following indications are provided:

- STI must be greater than 0.55 at normal speech level (60 dBA at 1 m distance) if the room volume is up to 250 m^3; and
- STI shall be greater than 0.50 for raised speech level (70 dBA at 1 m distance) if the room volume is greater than 250 m^3.
- STI must be greater than 0.60 for classrooms and auditoria equipped with a sound reinforcement system, whatever the room size.

Based on UNI 11532-2, the optimal values for C_{50} should be above +2 dB for rooms less than 250 m^3. If this criterion is met, the STI evaluation can be

superseded. For larger rooms, C50 is not applicable, and the STI value must always be assessed.
In terms of reverberation time, the optimal values for each building category are determined as a function of room volume, as shown in Figure 1. The acoustic classification of the buildings is indicated in Table 3.

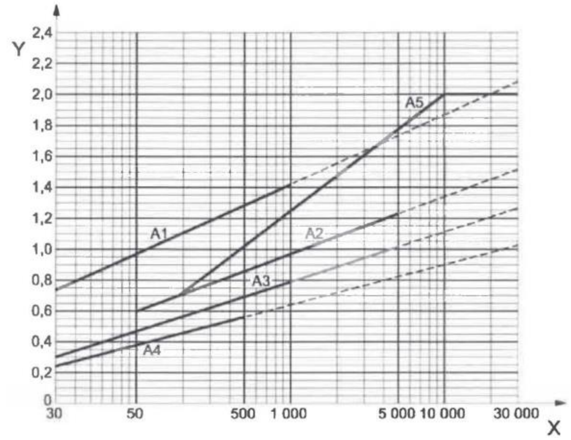

Figure 1. Optimal reverberation time at medium frequencies according to UNI 11532-2.

In Figure 1, the X-axis refers to the room volume expressed in m^3, and the Y-axis refers to the optimal reverberation time expressed in seconds.

Category	Room Function	Target
A1	Music	Good acoustics for non-amplified music
A2	Speech Conference	High level of speech intelligibility
A3	Lessons with student-teacher interaction	High level of speech intelligibility
A4	Lessons including special classrooms	High level of speech intelligibility with more speakers
A5	Sport, Gyms	Possible verbal communication for short distances
A6	Staircase, Library, Changing rooms	Noise control

Table 3. Acoustic classification of spaces in schools according to UNI 11532:2020.

For category A.6, no criteria are given for speech quality, only the maximum values for airborne noise generated by mechanical ventilation and activities within the room.
As previously mentioned, IEC 60268-16:2020 [18] is the standard for performing STI measurements. One of the innovations is that the rapid STI (RASTI) parameter, which was calculated only for the middle frequency bands between 500 Hz and 2 kHz, is no longer used. The complete procedure of STI measurements is described in section 4.2.
Predictive estimation of reverberation time in rooms can be performed according to ISO 12354-6:2003 [20]. The absorption in a room is described by the equivalent absorption area (A), as indicated in equation (7).

$$A = \sum_{i=1}^{n} \alpha_{s,i} \cdot S_i + \sum_{j=1}^{o} A_{obj,j} + A_{air} \qquad (7)$$

Where n is the number of surfaces, and $\alpha_{s,i}$ is the absorption coefficient of each surface element of the room, $A_{obj,j}$ is the absorption area of the j-th object, o is the total number of objects and A_{air} is the air absorption. The reverberation time is calculated as indicated in equation (8).

$$T = \frac{55.3}{c_0} \frac{V(1-\Psi)}{A} \qquad (8)$$

Where c_0 is the speed of sound in air and Ψ is the fraction of the volume occupied by objects (object fraction). A is the total equivalent absorption area, which includes absorption by internal surfaces, objects, and air. Eq. (8) is basically the Sabine's formula already introduced in eq. (1).
The limitations of ISO 12354-6 are that the calculation of reverberation time should be applied to regularly shaped volumes; absorption should be uniformly distributed. If these assumptions are not met, the reverberation time results may be higher than estimated.

4 Measurement techniques

There are two types of techniques and equipment for measuring the acoustic parameters of speech.

- The first technique uses an omnidirectional sound source and follows the sine sweep or impulsive source method, as described in ISO 3382-1:2009 [9]. This is used to measure T_{20} and C_{50}.
- The second technique uses a talk-box as a sound source, capable of reproducing the characteristics of the human voice in terms of directivity, spectrum and power level. This is used to measure STI according to IEC-60268-16:2020 [18].

4.1 Acoustical parameters according to ISO 3382-1

ISO 3382-1 describes methods for measuring the acoustic impulse response of a room and deriving a number of acoustic parameters from it. This standard describes the use of an omnidirectional source fed with MLS pseudo-random noise, or, better, with an exponential sine sweep (ESS) signal.
In the case of exponential sine sweep, the signal recorded by the microphone is convolved with the inverse filter of the signal emitted by the sound source to obtain the impulse response of the room.
In the post-processing phase, several acoustic parameters (including C_{50} and T_{20}, but excluding STI) can be obtained from the impulse response.
A very practical way to obtain all the parameters described in ISO 3382-1 at once is to analyze the impulse response using the "Acoustical Parameters" plugin, which is part of the Aurora software package [21]; Aurora is a collection of plugins running under Adobe Audition (up to version 3.0) and Audacity (version 2.4.1).
Aurora also provides tools for generating MLS or ESS test signals, and for deconvolving the impulse response from the recorded test signal.
Figure 2 shows the typical result obtained with the latest version 4.5c of Aurora Acoustic Parameter plugin.

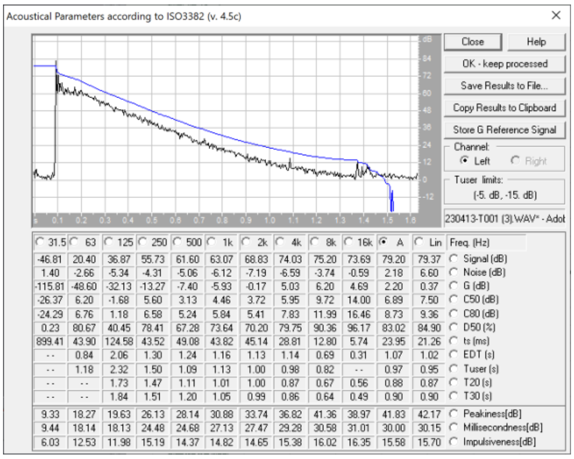

Figure 2. Acoustic parameters computed by the Aurora plugin according to ISO 3382-1.

4.2 STI according to IEC 60268-16

This second method uses an artificial human voice represented by a talk-box or a dummy head provided with an artificial mouth, which is fed with an amplitude-modulated test signal that must be calibrated to a sound pressure level of 60 dB(A) at 1 m distance from the center of the artificial mouth for normal voice level, or to 70 dB(A) at 1 m distance for raised voice [22].
For each octave band between 125 Hz and 8 kHz a carrier signal (pink noise) is amplitude-modulated at 14 modulation frequencies, ranging from 0.63 Hz to 12.5 Hz.
Based on gender, the spectra of STI can be normalized to an A-weighted sound pressure level of 0 dB according to Annex A of IEC 60268-16 [18], as summarized in Table 4. The difference between the 2011 and 2020 published versions of this regulation is that the latest version specifies the spectrum for male voice only; the female voice spectrum has been removed.
Table 4 compares the new "male" voice spectrum with the old "male" and "female" voice spectra.

Octave Frequency Band (Hz)	Male Voice (2020)	Male Voice (2011)	Female Voice (2011)
125	-2.5	2.9	-
250	0.5	2.9	5.3
500	0	-0.8	-1.9
1000	-6	-6.8	-9.1
2000	-12	-12.8	-15.8
4000	-18	-18.8	-16.7
8000	-24	-24.8	-18.0
A-weighted	0	0	0

Table 4. Human Voice Spectra.

The IEC-60268-16 [18] standard also describes an alternative method for measuring STI, in which the amplitude-modulated test signal is not used, with a separate RIR measurements and SNR, according to eq. (4).
A dedicated STI Aurora plug-in can be used for this alternative measurement procedure, which was recently updated to comply with the 2020 version of the IEC standard. In detail, four samples need to be recorded:

- The calibration signal of the microphone;
- The background noise level at the listening position;
- The pink noise signal equalized according to Table 4, emitted by the artificial mouth or talk-box and recorded at the listening position;
- The impulse response of the room measured with the talk box at full power.

Some details that complete the procedure are described below.

The calibration of the microphone is usually obtained fitting a calibrator on the microphone, which provides 94 dB at 1 kHz. The background noise is recorded at the position of the receiver where the measurement is to be made. After calibrating the emitted test signal (equalized pink noise, not amplitude-modulated) at 1 m distance, the emitted signal is recorded by the microphone placed at the position of the receiver in the room.
Then the impulse response is measured by feeding the talk box with a non-equalized exponential sine sweep signal and driving it to maximum power. The resulting room impulse response is essentially noise free, thanks to the high immunity to background noise provided by the ESS method.
The Aurora STI plugin is employed four times.

- At the first run, the SPL full scale value is calibrated by analyzing the recording of the tone generated by the calibrator and forcing the SPL value to be 94 dB.
- At the second run the octave-band spectrum of the background noise is computed and stored.
- At the third run the recorded test signal emitted by the talk box is analyzed, resulting in the octave band spectrum of Signal+Noise, which is stored (and automatically separated in the Signal and Noise spectra).
- Finally, at the fourth run, the RIR is processed. From it, the MTF values are computed by the Aurora STI plugin to obtain the overall STI value at that specific position in the room, according to equation (4).

The resulting MTF curves are also displayed, as shown in Figure 3.

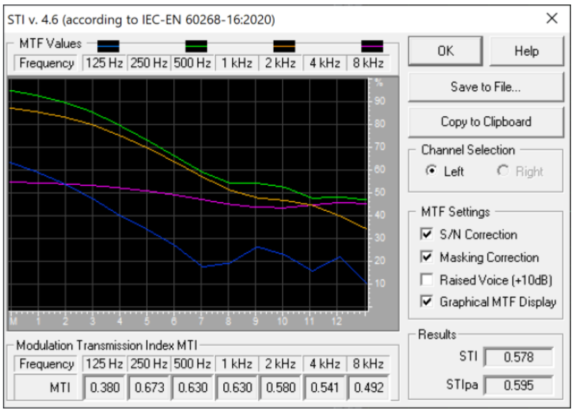

Figure 3. STI result using the new Aurora STI plugin.

The big advantage of using this indirect method is that it allows to evaluate simultaneously the STI value with the standard voice level and with the raised voce level, without the need of performing a second measurement.

5 STI calculation with Ramsete

Ramsete is a beam tracing program making use of triangular beams (pyramids) [23].
Ramsete has been recently updated for addressing the new standard IEC 60268-16:2020 [18]. The new version contains two calibrated sound sources, one is a true human talker where the directivity of the human voice has been calculated based on the measurements taken by Chu in 2001 [24] and by Farina in 2003 [25]. The second source contains the directivity data taken from the CLF file made available by the manufacturer for the Bedrock talk box (BTB-65) [26].

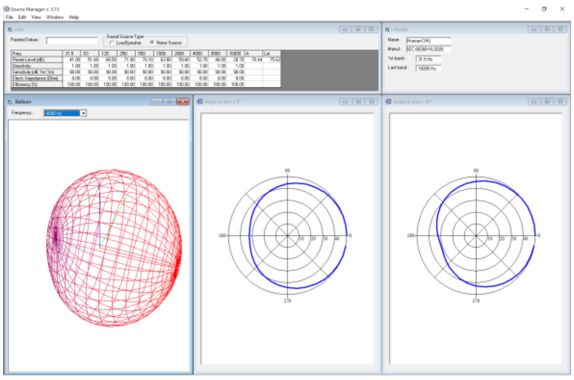

Figure 4. Directivity of the new “Human CHU” sound source at 4 kHz.

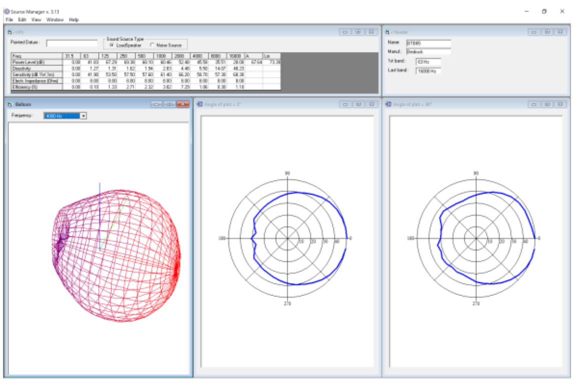

Figure 5. Directivity of the new “Bedrock BTB-65” sound source at 4 kHz.

Figures 4 and 5 show the directivity curves of the new calibrated sound sources: it can be clearly seen that the real directivity of human speaker (Fig. 4) is different from the directivity of a standardised talk-box (Fig. 5) despite both are within the limits for artificial mouths specified in the ITU Recommendation P51 [26].

For both sound sources the power level spectrum has been adjusted so that SPL equals 60 dB(A) at 1m in the front, and the spectrum corresponds to the new "male" spectrum described in IEC 60268-16:2020 [18] and shown in Table 4.

6 STI(ϑ) evaluation with MIMO IRs

In Ramsete the simulation is not performed in most cases with a specific sound source with fixed directivity and precise aiming. The latest version of Ramsete (currently 3.13) allows the RIR to be computed using a MIMO approach, where both the sound source and the receiver are multichannel transducer arrays, that can be "beamformed" to provide arbitrary directivity patterns.
Similarly, a MIMO room IR can be measured using a suitable spherical loudspeaker array and a spherical microphone array.
A common approach is to use High Order Ambisonics (HOA) arrays, where both the source and receiver provide a set of predefined directivity patterns, described by spherical harmonics up to a specific order (usually 3rd order, which means 16 patterns).
An alternative approach is to use the Spatial PCM Sampling (SPS) to define a number of directive transducers at both source and receiver.
This MIMO approach to creating matrices of room impulse responses with HOA or SPS patterns is described in [28].
After obtaining a MIMO impulse response matrix, either from measurements using Aurora or from calculations with Ramsete, it is possible to evaluate STI by setting the sound source to emulate a human speaker facing a given direction, and beamforming the receiver to become a perfect omnidirectional microphone.
When the MIMO matrix is a HOA 16×16 matrix (third order), a 1×16 beamforming matrix must be applied at the source side, and only the first of the 16 output channels, corresponding to the Ambisonics pattern "W", which is omnidirectional, must be analysed.
Two 1×16 beamforming filter matrices were created, using Excel's "solver" function for estimating the gains to be applied to each Ambisonics component for modelling the directivity balloons of the "Human Chu" and "Bedrock BTB-65" sound sources.
Figure 6 compares the directivity curves on the horizontal and vertical planes of the "Human Chu" sound source with its Ambisonics 3rd-order model at 4 kHz.

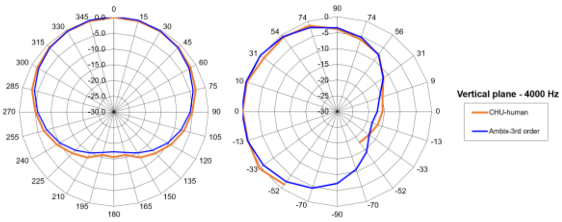

Figure 6. Ambisonics model of the directivity of the "Human Chu" sound source at 4 kHz.

The "Human Chu" filter matrix for Ambisonics MIMO IRs is included in the Beta version of Ramsete 3.13.
The availability of these FIR filter matrices allows to obtain the room impulse response generated by a human voice, and hence use it to compute STI according to the indirect method described in the previous subsection, using the Aurora STI plugin.
This approach can be applied to MIMO IRs computed by Ramsete or measured with transducer arrays described in [27].
As an Ambisonics-modelled source can be easily rotated in any direction, this approach can be used to study how the STI value changes when the talker is not facing the listener, but is rotated in a different direction.
As an example, Figure 7 shows the STI(ϑ) value as a function of the azimuth angle ϑ where the source is facing. It was obtained by reprocessing a MIMO IR measured in Teatro Municipale of Piacenza [29] for a receiver located in the 7th row of seats (receiver No. 2), directly aligned with the sound source on stage (source L).

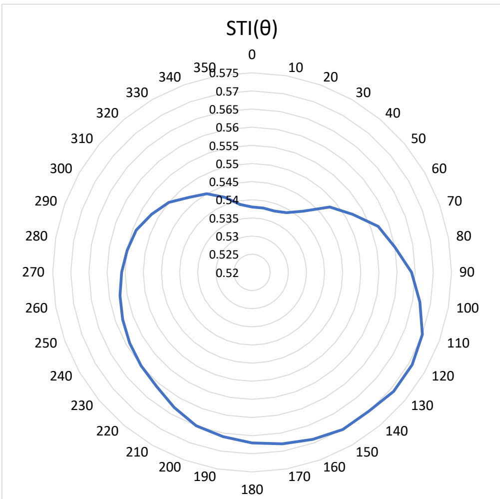

Figure 7. STI as a function of azimuth angle ϑ of aiming the sound source.

It can be seen how STI increases slightly when the source is aimed backwards (towards the audience).

7 Conclusions

In Italy the environmental acoustic parameters required for public buildings, and more specifically for schools, have been revised following the release of the new decree MITE No. 256 of 23/06/2022.
As a result, the software used for acoustic measurements and simulations had to be updated to comply with the new law and the current version of the ISO and IEC standards, including the new "male" spectrum for STI measurements. The new software also includes accurate models of the directivity of human talkers and a commercial talk box. These directivity data have also been transformed in FIR filter matrices, which can be used for beamforming a High Order Ambisonics source, employed for MIMO IR measurements or calculations.
Thanks to these developments, the value of STI can now be computed or measured as a function of source orientation.
Future research will include a comparison of some case studies of speech intelligibility, employing both real measurements and digital simulations.
Measurements will be performed using both the direct method (amplitude-modulated signal automatically generated by the talk box) and the indirect method (separate measurements of noise, signal and impulse response).
The MIMO IR approach will be compared to results obtained with a real talk-box and a real omnidirectional microphone to assess the capability of the MIMO IR approach of measuring STI(ϑ), depending on the orientation of the talker, with a single MIMO measurements, removing the need of repeating the measure while the talk-box is rotating.

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