Speech intelligibility measurements in practice
Obtaining accurate and reliable data using your Bedrock meter

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Speech intelligibility measurements in practice;
Obtaining accurate and reliable data using your Bedrock meter

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1. Introduction

Measuring speech intelligibility using the Speech Transmission Index method is not particularly difficult. With the proper tools and a minimal amount of practice, measurement results are quickly obtained that are the most accurate among all intelligibility test methods. That being said: there *are* some pitfalls to be aware of, and mistakes that undermine the validity of measurements *do* occur – not just with not first-time users. There is an abundance of papers and other information resources concerning the Speech Transmission Index available [1,2], but almost all of this is intended for developers and expert-level users. We admit that this is especially the case with the papers and books published by us (the authors of this white paper). This is our effort to fix this omission, and provide a tutorial that covers (only) the basics of STIPA and Full STI testing. If some of the steps we are describing here seem trivial, we apologize for this in advance. For the purposes of this white paper, we prefer repeatedly stating the obvious over forgetting crucial steps.

> *If you are truly new to intelligibility testing, or perhaps even new to acoustic testing procedures altogether, we recommend that you watch some introductory "explainer videos" on the Bedrock website: http://www.bedrock-audio.com/explainer-videos*

In the next section op this paper, we will start by definining Full STI and STIPA, and how these methods are used to measure speech intelligibility. In section 3, we will provide an overview of the tools you need for doing Speech Transmission Index measurements. Section 4 continues by suggesting a generic set of "standard operation procedures" to adopt for STI measurements, followed in section 5 by specific details on the procedures to be followed for NFPA72 Annex D (being a common example of STI testing procedures). In section 6, limitations of the Speech Transmission Index method in general (and STIPA in particular) are described, followed by a few more advanced processing options in section 7.

> *This paper covers the basics of speech intelligibility testing. For information on the physics and mathematics underlying the Speech Transmission Index, please refer to the publications cited in the list references included with this paper, in particular the book "Past, Present and Future of the Speech Transmission Index." This book is available for download, free of charge, from the Embedded Acoustics website: www.embeddedacoustics.com [2]*

2. What exactly is the Speech Transmission Index, and how does it work?

2.1. Origins of the STI

The Speech Transmission Index (STI) model is a framework for predicting speech intelligibility that was originally invented by Steeneken and Houtgast in the late 1970s [3]. It builds on earlier work done by other researchers (mostly at Bell Labs) that resulted in the Articulation Index (AI), which was the dominant model into the 1960s. Steeneken and Houtgast introduced a new conceptual approach that was inspired by optical reseach, based on the so-called Modulation Transfer Function. This led to some very valuable and unique characteristics of the STI over other metrics:

- The STI predicts the influence of virtually all influences that degrade intelligibility, not just those reflected in the speech-to-noise ratio. This includes reverberation, echoes and nonlinear distortion.
- The STI can be measured directly with a deterministic test signal. There is no requirement to do separate measurements for speech and noise (which is usually not feasible in practice).

While a full theoretical explanation of the STI-model is beyond the scope of this white paper, a brief explanation of how the model works is given in section 2.5 of this white paper. Described in the most general terms: the STI tracks if the naturally occurring modulations in speech are being preserved while speech is being transmitted through a channel. Steeneken and Houtgast observed that if these modulations are lost, then intelligibility is decreased proportionally. This observation holds for the vast majority of degradation sources.

> *While the STI was originally proposed back in the 1970s, it is still very much up-to-date. As authors of this paper, we are committed to continue the research needed to keep the STI up to speed with new technology and developments. There is an active community of researchers that support the STI, as well as a standing IEC maintenance committee.*

2.2. Difference between STIPA and Full STI

In theory, there are several variants of the Speech Transmission Index to be chosen from, tailored for different applications[1]. In practice, the choice is limited. The original version of the STI (now mostly referred to as "Full STI") is the most universally applicable version: it is suitable for virtually any kind of transmission channel and application. No commercial implementations of the Full STI were available between 1998 and 2018. The Full STI ended up being disused until Bedrock Audio re-introduced it in 2018. The fact that a single measurement took approx. 15 minutes reduced its appeal considerably. The re-introduction was prompted by the development of new signal processing techniques (by the authors of this paper) that reduces the measurement time to 70 seconds.

Several other (now obsolete) STI variants, such as RASTI and STITEL, were popular through the 1990s, despite known limitations of these simplified versions of the STI. The introduction of STIPA around 2000 pushed all these other variants to the background. The only two STI implementations of relevance today are Full STI and STIPA.

STIPA stands for "Speech Transmision Index for Public Address systems." It is simplified implementation of the Speech Transmission Index model, in the sense that it considers only 2 modulation frequencies per octave band instead of the full range defined for the Full STI. However, when applied to public address systems, the impact of this simplification on the accuracy of the measured STI is negligible. As it turned out soon after STIPA was introduced, the same is true for many other applications, including most room acoustics studies. STIPA quickly became popular for all kinds of applications, not just PA systems. In just a few years STIPA almost completely replaced all other STI variants, simply because it was comparatively fast (~20s per measurements) and it turned out to be nearly universally applicable [4]. STIPA became almost a synonym to the Speech Transmission Index, and its limitations (compared to the full STI were disregarded).

However, those limitations are still there, and should be considered before you select STIPA for your application. In particular, STIPA is not suitable if one of the following two conditions occurs:

- Distinct echoes occur (in very large spaces)
- There is a combination of severe reverberation and bandwidth limiting (e.g narrow-band PA systems in reverberant rooms).

In 2018, when the authors of this paper re-introduced Full STI based on novel signal processing algorithms, it was immediately implemented it in the Bedrock product line. The processors in the Bedrock meters are powerful enough to support the novel algorithms, that allows simultaneous measurements of all modulation frequencies and extends upon the approach already adopted for STIPA to measure the Full STI. A full STI measurement now takes 70 seconds, which is considerably longer than STIPA, but short enough to be feasible for most purposes (including larger field measurement campaigns).

The limitations of STIPA and the Full STI are further addressed in section 6 of this paper.

> *STIPA and full STI are used to objectively measure speech intelligibility, using a specific test signal (the STIPA or Full STI signal) in combination with a specific, dedicated acoustic measuring instrument (the STI analyzer, such as the Bedrock SM50 or the Bedrock SM90)*

Although it is good to be aware that other STI variants have been used, and are still mentioned at times, we would like to stress that all variants other the Full STI and STIPA are currently considered obsolete and should not be used. Where standards specify RASTI or STITEL, it is usually considered acceptable to use STIPA or Full STI instead.

Table 1 summarizes the key differences between STIPA and Full STI

Table 1. Key differences between STIPA and Full STI

	Full STI	*STIPA*
Suitable in case of echoes (e.g. cathedrals)	yes	no
Suitable when a combination of bandwidth limiting and reverberation occurs	yes	no
Suitable for PA systems	yes	yes
Suitable for all other applications in electro-acoustics and room acoustics not mentioned above	yes	yes
Measuring time	70 s	15-25 s
Modulation frequencies per octave band	14	2

Each Speech Transmission Index results in a 0 – 1 index, the STI. In written reports the correct way to refer to results from STIPA or Full STI measurements is as Speech Transmission Index (or STI) values, and not as STIPA or Full STI values. STIPA and Full STI are the methods you use to obtain STI results.

2.3. The STI characterizes speech transmission channels

Although it is nowadays generally accepted to loosely refer to STI measurements as “speech intelligibility measurements,” this is not completely accurate. The Speech Transmission Index reflects how a transmission path *affects* speech intelligibility; it is a purely physical measure that does not take listeners and talkers into account, but it just characterizes the transmission channel. This means that factors such as hearing loss, poor articulation and other (human) limitations are not taken into account. This sounds like a limitation of the method, but in practice, this is usually a good thing. If you are the supplier of a PA system that is being certified using STIPA, you do not have to worry about the sloppy speech style or the impaired hearing of the evaluators (or other factors out of your control) to affect the outcome of the tests. Standards usually set performance limits based on the STI under the (often implicit) assumption that all talkers and listeners are “normal.” This has one potential drawback: it means that expectations based on STIPA measurements may be too optimistic if, in practice, large populations of hearing impaired people need to be addressed, or announcements are made in an accented voice or with a sloppy speech style. In those cases, performance limits need to be set to higher STI values to guarantee sufficient intelligibility. IEC-60268-16 gives guidance on how to adjust STI requirements in such cases.

To summarize: the STI tells you only what the *speech transmission channel* does to the speech in terms of intelligibility. Before anything else, you need to consider what the speech transmission channel you intend to test comprises. The term speech transmission channel in the context of STIPA testing is used in a broader sense then (for instance) in telecommunications engineering. The term “channel” suggests to some that electronic equipment (e.g. for radio transmission) is used, which is not necessarily the case. Figure 1 shows the definition of the speech transmission channel: basically everything that influences intelligibility, except to the talkers and listeners themselves.

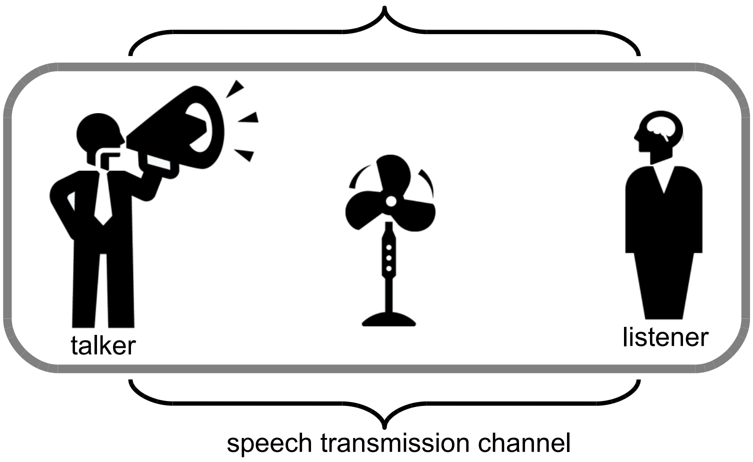


Figure 1. Schematic representation of the definition of a speech transmission channel. The channel comprises everything between the talker and listener that influences intelligibility, including noise sources and the acoustics of the environment, except for the talker and listener themselves.

In fig. 1, the fan symbolizes a noise source interfering with speech from the talker. The talker and listener are occupying the same space, the acoustic properties of which (determined by wall materials, ceilings, etc.) will have an influence on intelligibility. This is also taken into account by the STI. The bullhorn used by the talker represents the use of electro-acoustic devices. Such devices, if present in the transmission channels, may introduce non-linear distortion components which are also taken into account by the STI.

> *Transmission channels tested using the STI method may include combinations of any of the following factors that influence intelligibility: ambient noise, room acoustics (reverberation, echoes), electronic noise, non-linear distortion, amplification, filtering (modifying the frequency transfer function). Explicitly NOT included in the test results are all factors related to the talkers and listeners themselves (accent, hearing loss, etc.)*

2.4. Speech intelligibility versus speech quality

Speech intelligibility is a measure of the effectiveness of speech communication. Speech intelligibility is usually defined as the percentage of speech units (syllables, words or sentences) correctly perceived by listeners. Formal speech intelligibility measurements are usually done with panels of listeners and calibrated speech signals. In a tightly controlled test setup, participants (listeners) repeat back exactly what they hear. This is compared to the source material to precisely compute the percentage perceived correctly. If done properly, these subjective tests are by far the most representative way to measure intelligibility, because no assumptions are made at all: speech intelligibility is measured directly according to its definition. Even so, subjective intelligibility tests are hardly ever done these days, simply because the equivalent of a 25-second STIPA test would involve at least 2 subject-hours of listening tests, and about 30 minutes of recording speech samples from the channel under test. Whereas a STIPA test instantly shows the results (the STI value), a subjective test has to be completed before the final results are known – which usually takes a few weeks.

> *Perhaps the most common misconception about speech intelligibility is that a trained listener can judge speech intelligibility by simply listening to a few sentences.* ***No accurate statements about speech intelligibility can be made by just listening to random speech materials for a few minutes.*** *It is, of course, possible to get an impression of how severely the speech is degraded – but this does not immediately tell you the effect on speech intelligibility.*

It is impossible to listen to speech without forming an opinion on the *quality* of the speech signal. It is important to keep in mind that speech quality ("how good does it sound?") and speech intelligibility ("how much can I understand?") are not the same measures. The distinction is not just academic. Nearly all forms of non-linear degradation, such as peak clipping, have a severe adverse effect on speech quality, whereas speech intelligibility sometimes even increases!

Obtaining a reliable subjective impressions of speech intelligibility in the field, independently from STI tests, is usually not at all feasible. Keep in mind that subjects memorize words quickly, and word lists can therefore never be re-used. This effect is much worse still with sentences, which are really not suitable for evaluation speech intelligibility in the field.

> *Remarks made about speech intelligibility based on personal observations should really be interpreted as opinions on speech quality.*

2.5. How does it work? Why is the STI test signal representative of speech?

Now that we have defined what a transmission channel is, and that we have established the difference between speech intelligibility and speech quality, the next question to address is: *how* is the STI used to measure how a speech transmission channel affects speech intelligibility?

To perform any STI test, the following three steps are taken:

- Replace the (human) talker by a source of the STIPA or Full STI test signal. Natural speech cannot generally be used for the test; it has to be a test signal designed specifically for the purpose of STI testing. This needs to be introduced in the same way as the natural, human speech would be (e.g., using the same microphone, at the same distance, etc.)
- Make sure that the transmission channel is controlled and stable, and representative of the scenario you wish to test. For instance: to test the PA system in a mall, make sure that the PA system is functioning as it normally would, with the settings you consider to be representative of normal operation, and with a representative level of background noise.
- "Replace" the human listener by the microphone of the STI analyzer (e.g. a Bedrock SM90). In practice, the STI analyzer is usually moved around the test venue to obtain readings at multiple locations. Since a single measurement takes only 25 seconds, it is not uncommon to do multiple measurements at each location (to be able to analyze the statistics later on), and to perform up to hundreds of individual measurements for a single venue.

The STI signal is a noisy signal that contains all frequencies present in human speech, from approx. 80 Hz to 11 kHz. A common misconception, probably triggered by age-old conventions in telecommunications engineering, is that human speech is limited to the range between 300 and 3400 Hz. The actual range is much wider, and the entire range is taken into account by the STI.

An STI signal sounds nothing like speech. Yet similar to speech, it features modulations (intensity fluctuations) in the range from 0.6 to 12.5 Hz. In real speech, intensity fluctuations form the main mode used to encode information. Loss of these modulations means loss of information. This is how the STI works: the loss of these modulations is measured. Instead of pseudo-random fluctuation patterns, deterministic patterns are used that make it easy for the analyser to measure the modulation depth. For STIPA, two modulation frequencies are considered per octave band; for full STI, 14 modulations per band are measured.

> *In real speech, information is "encoded" through intensity fluctuations. Loss of modulation depth means loss of information, which translates into a reduction of speech intelligibility. This is how the STI works: the artificial STI test signal) has a carefully designed pattern of modulation frequencies. The STI analyzer measured how the transmission channel has reduced the modulation depth. This is measured in multiple frequency bands (125 Hz – 8 kHz), across a range of modulation frequencies (0.6 – 12.5 Hz).*

2.6. Interpreting STI values.

The Speech Transmission Index is a single-number rating in the range between 0.00 and 1.00. Obviously, zero corresponds to not intelligibility at all; a value of 1.00 means perfect intelligibility. Note that even at an STI of 1.00, speech *quality* may be perceived as far from perfect.

The question is: how do we interpret the scale between 0 and 1? The original inventors of the STI, Steeneken and Houtgast [3], introduced the following table to translate the STI values into meaningful categories.

Table 2. STI category labels

STI category label	*STI range*
bad	0.00 – 0.29
poor	0.30 – 0.44
fair	0.45 – 0.59
good	0.60 – 0.74
excellent	0.75 – 1.00

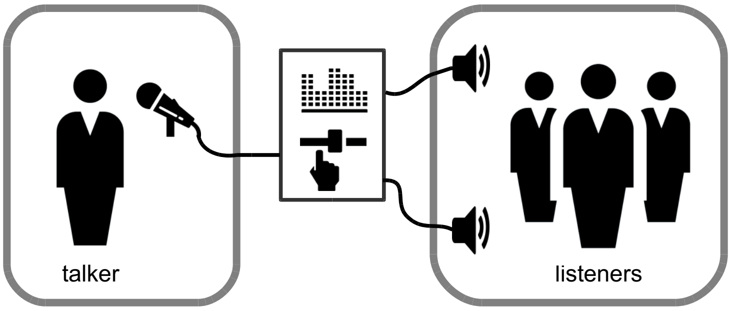
Application standards introduce acceptance limits based on the above table. Mininum values commonly required by application standards are 0.45 and 0.50. In practice, any transmission channel that shows STI values below 0.35 is not fit for any kind of use. What limit is appropriate depends on the type of use to which the speech transmission channel will be applied. Performance limits are usually raised if large proportions of the listener population are expected to be hearing impaired or non-native listeners.

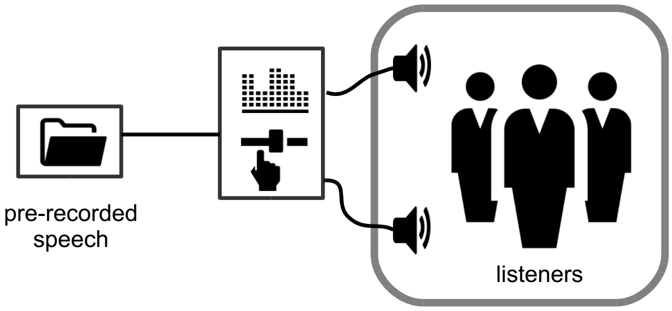
STI values always rounded to two decimals. The uncertainty associated with a single STIPA measurement is 0.02 to 0.03. A higher accuracy can be achieved by taking the average across multiple measurements.

2.7. Examples of different types speech transmission channels to be tested with STIPA or Full STI.

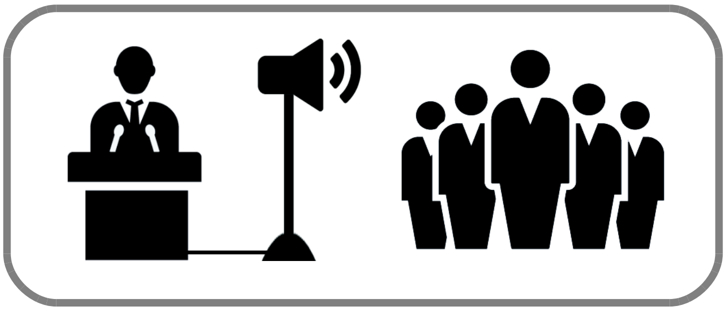
The most common application of the STIPA test method is to evaluate PA systems – hence the "PA" in STIPA. Below, we present a few common STI testing scenarios, for which we explicitly define what the transmission channel to be tested is, what we consider to be our talkers and listeners, and what factors we expact to be of influence on the STI. Keep in mind that during STI tests all talkers will be replaced by a source of the STI test signal and listeners (and listener locations) will be measuring positions where the STIPA analyser will be used. For the scenarios introduced below STIPA and Full STI are equally suitable.

Scenario 1 is the typical PA and Voice Evac scenario. Scenario 2 is similar, the difference being that pre-recorded speech is used instead of a live talker. Scenario 3 is the "classic" application of the Speech Transmission Index to pure room acoustics, without the involvement of electronics for sound reproduction. STIPA evaluations can be very useful in identifying the impact of factors relating to room acoustics (e.g. lack of acoustic absorption materials) and ambient noise (e.g. due to airconditioning systems) on speech intelligibility. Scenario 4 is a typical lecture-type situation, where one lecturer speaks to a larger number of listeners in the same room.

Scenario 1. ***PA with "live" announcer***	
Talker	The talker is usually a single individual making announcements, who may (or may not) have been trained to this purpose. The talker is usually is out of reach of the PA and unable to hear his/her announcements back.
Listeners	Listeners are all the people present in the venue for whom the announcments are intended. Listener locations to be considered are all spaces where the public is allowed.
Speech transmission channel to be tested	Everything from the paging microphone (in its acoustic environment) up to all relevant listener locations. This includes PA electronics.
Factors of influence on the STI	• Noise and reverberation at the talker location, • Paging microphone characteristics and speaking distance • Frequency transfer modification and distortion due to PA • Noise and reverberation at the listener location • Overall sound pressure level due to PA

Scenario 2. ***PA with pre-recorded announcements***	
Talker	Instead of a live talker, "canned" speech is used (usually high-quality studio recordings by professional voice actors).
Listeners	Listeners are all the people present in the venue for whom the announcments are intended. Listener locations to be considered are all spaces where the public is allowed.
Speech transmission channel to be tested	Everything from (digital) audio storage and playback electronics up to all relevant listener locations. This includes PA electronics.
Factors of influence on the STI	• Frequency transfer modification and distortion due to PA • Noise and reverberation at the listener location • Overall sound pressure level due to PA

Scenario 3. "Live" meetings and conversations	
Talker/listeners	In meetings and conversations, the same people take turns acting as talkers and listeners. All positions around a meeting table are therefore to be considered as talker positions as well as listener positions
Speech transmission channel to be tested	Each individual talker and listener position combines into a transmission channel.
Factors of influence on the STI	• Distance between talker and listener • Revererberation in the meeting room • Ambient noise in the meeting room; interfering speech from adjacent rooms • Vocal effort; speaking levels (relaxed vs. raised voice)

Scenario 4. Lecture	
Talker	A single lecturer usually addresses a room full of people. The talker position will be at the lectern, using a fixed microphone, or a somewhat larger presentation area if a wireless microphone is used.
Listeners	All seats in the audience are seen as listening positions. Generally there are more seats than can realistically be covered by STIPA measurements. A selection of representative seats (which must always cover the expected worst-case seats) is to be selected.
Speech transmission channel to be tested	Everything from the microphone up to all listener positions in the room.
Factors of influence on the STI	• Noise and reverberation in the lecture hall • Microphone characteristics and speaking distance • Frequency transfer modification and distortion due to the sound system; possible the influence of feedback • Overall sound pressure level generated by the sound system, which will differ from seat to seat.

The four scenarios are the most common STIPA testing applications, but this is definitely not an exhaustive list. For instance, the Speech Transmission Index has also proven its value for evaluating just parts of the transmission channels shown in each of these scenarios, such as just a type of loudspeaker or microphone.

Full STI measurements are typically used in the following settings and scenarios:

- During formal certifications, out of an abundance of caution, Full STI is sometimes used instead of STIPA even if STIPA technically meets all requirements. It increases the net measuring time by a factor of 3-4, but given that most time usually goes into setting up, travelling to a venue, writing the report, etc – the actual measuring time is not always a significant factor to consider.
- Whenever there may be the slightest doubt if STIPA might be inaccurate (e.g. long reverberation times, slightly out-of-phase loudspeakers, etc), it is good practice to at least measure Full STI for a (small) subset of all test conditions, and compare the results to STIPA. If Full STI and STIPA correspond well (<0.03 difference between any two individual measurements), then it is safe to use STIPA.
- Whenever there are echoes (either naturally occurring or because of out-of-phase loudspeakers), or if there is a combination of long reverb times (>3s) and bandwidth-limited sound systems, then Full STI is required and STIPA results will certainly be less accurate.
- Sometimes STI measurements are matched up with STI predictions that were obtained with acoustic simulation software. Modelling and simulation packages typically implement Full STI and not always STIPA. Measurements done to validate predictions are usually done with Full STI if the predictions are also based on Full STI.

3. Equipment and resources needed for an STI measurement

3.1. Availability of the transmission channel

To be able to do an STI test, first of all you need to have access to the transmission channel you aim to evaluate. This is not always as easy as it sounds, since you will probably be playing back STI test signals at high sound levels for several hours. For a PA system in mall (which is the example we used before, and which corresponds to scenarios 1 and 2), performing tests during opening hours is usually not an option. STI signals are not pleasant to listen to. Measurements will therefore have to take place after hours, when the mall is empty. However, the level of background noise, which is largely dependent on the number of people in the mall, will not be realistic. In other words: you don't have access to the transmission channel under realistic conditions. There is a common workaround for this problem, which comes down to measuring the background noise during the day, and measuring STI at night. The background noise is then computationally taken into account in the STI calculation. This procedure is explained in section 7 of this paper.

3.2. STIPA or Full STI test signal source

It is not possible to do STI measurements without a source of the proper STI test signal (either STIPA or Full STI).

> *STIPA tests can only be done based on the STIPA test signal compliant to IEC060268-16 rev. 3 or higher [5]. Full STI measurements require a test signal compliant to IEC060268-16 rev. 4 or higher.*
> ***Natural speech or other test signals cannot be used.***

STIPA signals compliant to the IEC-standard are available from all manufacturers of STIPA measurings solutions. The Embedded Acoustics signal is available for download as WAV-file or as an MP3-file from www.bedrock-audio.com, and can be used free of charge for all purposes except redistribution. Full STI tests signals can also be downloaded from the Bedrock site. Full STI is currently only supported by Bedrock meters.

Ideally, a real human talker would act as a STI signal source. Given the fact that for obvious reasons this is impossible, there are 4 ways to introduce the STI test signal.

STIPA or Full STI Signal Source option 1 ***Directly using a STIPA or Full STI file***	
When is this the best choice?	If the transmission channel under test uses pre-recorded speech, then replacing pre-recorded speech files by the STIPA of full STI signal file is the best and most representative option (**scenario 2**)
Advantages	No additional signal playback hardware is needed for the test
Drawbacks	If playback of a pre-recorded signal matches the type of use you wish to evaluate the channel for, then there are no drawbacks
Precautions	You need to make sure that the signal file is played back at the appropriate signal level. You may need to adjust the standard signal file using a wave-editor to achieve the proper correct level, and/or you may need to adjust gain settings in the message playback system.

STIPA or Full STI Signal Source option 2 ***Electrical signal input***	
When is this the best choice?	If you wish to evaluate a transmission channel featuring electronic sound reproduction, but wish to leave the influence of microphones outside of the equation.
Advantages	No sound is audibly played back at the input of the system under test; testing is generally more convenient than measuring acoustically with a TalkBox.
Drawbacks	If the microphone has an impact on intelligibility (which is generally the case), then this influence on the over STI is being ignored. Also, signal level calibration requires attention.
Precautions	The signal level of the playback device must be matched to the signal level that would normally be generated by the microphone of the system under test.

STIPA or Full STI Signal Source option 3 ***TalkBox*** (Image: Bedrock BTB65 TalkBox. Note that this device also has a balanced XLR output for us as an electrical signal source as decribed in option 2)	
When is this the best choice?	Whenever a calibrated acoustic source of the STIPA or Full STI signal is needed. This is true in scenarios 1, 3 and 4 as described above, and almost every other STI testing application.
Advantages	If you use a precalibrated TalkBox, then setting up is easy, and there is no need for level calibrations. Simply put the talkbox in the position relative to the microphone where talker's mouth would usually be.
Drawbacks	"Homebrew" talkboxes, although the most inexpensive tools that one can use as a STIPA signal source, are very difficult to calibrate. Not only the level has to match the (human) reference levels are given by the IEC-standard, but also the spectrum.
Precautions	The exact distance and alignment of the microphone relative to the talkbox is critical, especially for microphones meant to be used close to the mouth. The difference between 20mm and 40mm distance is 6 dB, which can correspond to major differences in terms of STI.

STIPA or Full STI Signal Source option 4 ***Head and Torso Simulator (HATS)*** (Image: Embedded Acoustics STIPAhats, a specialty tool for measuring the STI with SCBA face masks)	
When is this the best choice?	Whenever the shape of the human head has a profound impact on the outcome of the STIPA tests. This is the case when testing face masks and certain types of headsets.
Advantages	The closest match to a real human talker technically achievable.
Drawbacks	Moderately to very expensive, heavy, difficult to calibrate.
Precautions	Make sure that a HATS is truly what your application requires. If so, precisely follow the procedures for using the HATS (usually explained in detail in an application standard).

The words “level calibration” came up for signal sources 1 and 2. Whereas talkboxes and HATS’s generally make sure that the STI test signal level corresponds to “normal” levels of human speech, this is not the case when using electronic playback devices and signal files. If you use a STI test signal file or sound player, then it is your responsibility to match the signal level to the level of speech normally played back.

The procedure to match the STI test signal level to the level of the speech messages is (in general) as follows:

- Measure the A-weighted equivalent continuous Sound Pressure Level of a representative set of voice messages. This type of measurement, indicated as LAeq and sometimes also called the time-averaged A-weighted level, is measured by every kind STIPA analyzer.
- When selecting voice messages to base the LAeq measurement on, make sure that all natural pauses (between words) are included, but avoid unnaturally long pauses between sentences.
- LAeq is essentially the level of your spoken messages to be matched by the STIPA signal. But there is a catch: the STIPA signal is fully continuous, whereas real speech contains natural pauses, as mentioned before. Simply matching the level of the STIPA signal to the speech level result in a level that is too low – adjustment for the pauses is needed. IEC60268-16 rev.4 (section 5.1) [5] states that a correction factor of 3 dB must be applied. In other words: make sure that the LAeq of the STIPA signal is 3 dB higher than the LAeq you measure for a representative selection of spoken messages.

> *In cases where the STIPA signal is generated electrically or played back from a file, matching the level of the signal to the expected speech level is absolutely essential. Matching takes place based on the equivalent-continuous (aka time-averaged) A-weighted Sound Pressure Level.* ***The LAeq of the STIPA signal has to be set 3 dB higer than the (average) LAeq of a set of representative spoken messages.***
>
> The 3 dB adjustment for pauses is *not* needed when measuring the speech level with the Speech Level Meter module featured on Bedrock meters. This module automatically compensates the level for pauses in the signal. This is a more accurate approach than compensating by the (average) correction factor of 3 dB.

If you use a calibrated talkbox or HATS, these calibrations are generally not necessary. These devices are pre-calibrated to match the source power and directivity of a human speaker. However, it is good practice to always verify that the talkbox does indeed produce the correct level.

The level produced by human talkers may vary somewhat; people tend to speak a bit louder during presentations than during relaxed conversations in a quiet environment. The term "vocal effort" is used to indicate the A-weighted sound pressure level at a distance of 1 meter from the mouth. Standardized values for the vocal effort, as used in the STI-standard IEC-60268-16 [5], are defined by ISO-9921 [6]. This standard defines that "normal" speech corresponds to a vocal effort of 60 dB(A) measured at 1 meter distance. "Relaxed" speech is defined at 54 dB(A), and "raised" as 66 dB(A). A further series of levels is defined in 6 dB steps, up to "maximum shout" at 90 dB(A).

> *Based on ISO-9921, IEC-60268-16 rev. 4 states that the reference speech level from a talkbox has to be* ***60 dB(A) measured at a distance of 1 meter****. Always use this value as your baseline level, unless you are dealing with an application standard that explicitly contradicts the STI standard and calls for a different reference level (such as NFPA72). It is common practice to investigate the influence of vocal effort variations by adjusting the level in 6 dB steps, from 54 dB(A) to 72 dB(A). Levels measured at 1 meter distance higher than 72 dB(A), or at most 78 dB(A), are not to be considered realistic in most scenarios.*

Note that is usually not practical to verify the talkbox level by literally measuring the level at a distance of 1 meter nor is it a practical paging distance between talkbox and microphone. At this distance, the acoustics of the room may have a significant disturbing influence on vocal effort measurements. Instead, it is recommended to verify the talkbox level settings at distances of 0.50 and/or 0.25 meter, correcting for the difference in distance. Furtermore, a pratical working distance between the talkbox and microphone has to be established.

> *The reference level of 60 dB(A) at 1 meter distance corresponds to 66 dB(A) measured at 0.50m and 72 dB(A) at 0.25m. The level decreases by 6 dB whenever the distance is doubled.*

3.3. STIPA or Full STI analyzer

We have discussed how the STI test signal source replaces the talker. Similarly, the STI analyzer replaces the listener. The difference is that there is usually only one talker, at a fixed position, whereas in most scenarios there are many listeners who have the freedom to move around. This means that many measurements are usually done in a variety of locations, while the operator moves the STI analyzer from place to place. Fortunately, these days STI analyzers are compact handheld-devices capable of operating on battery power.

Figure 2. The Bedrock SM50 STIPA meter [7]. Whereas this instrument was originally developed as a STIPA meter, and has basic as well as more advanced STIPA features, it also includes other acoustic measurement modules, such as a real-time spectrum analyzer, RT60 meter and a sound pressure level meter.

A Speech Transmission Index analyzer generally consists of the following components:

- A microphone. The STI standards that the microphone has to comply with class 2 / type 2 specifications, or better.
- An internal pre-amplifier for the microphone signal.
- A central signal processing unit, that computes the STI from acoustic signals.
- A display device
- Buttons and/or touch screen to control the device
- Batteries or external power source.
- A USB or serial connector to download data to a computer

For general instructions on how to operate the STI analyzer, you will need to consult the manual of your device. This differs between device types and manufacturers. The basic principles remain the same across all STIPA measuring instruments.

When purchasing an STI analyzer, it is essential that the device complies with the latest standard. This is currently IEC-60268-16 rev. 5 (2020). Other important aspects to consider are easy-of-use (some user interfaces are very complex) and advanced processing options (such as computationally adding background noise to STIPA measurements).

4. Generic STI test procedures

4.1. Drafting a test plan

The first step for any STI-based evaluation is to draft a test plan. By drafting a plan ahead of the actual measurements, you make sure that nothing is forgotten once you actually run the tests, and you limit the (usually valuable) time needed at the actual test venue. This also gives you a chance to review the standards that apply to your type of testing in case the transmission channel under test has some unusual characteristics, or if you are dealing with missing our ambiguous specifications.

Input for the test plan usually comes from the following sources:

- The schedule of requirements for the speech transmission channel under test: what is the purpose of the channel, and what levels of performance are required?
- The applicable standards, which are usually cited in a schedule of requirements, but may also implicitly be applicable by law. IEC-60268-16 applies to STIPA tests, but this only describes the test methodology itself. Application standards, often set at the national level, further define exactly *how* and *where* you need to do STIPA tests. Examples of such application standards are NFPA72 Annex D (USA) and NEN2575 (Netherlands). Description of all application standards is beyond the scope of this paper, since there are many different standards which are also changed quite often.
- Additional (local) regulations. Sadly, it is not uncommon that very specific rules apply at the level of the individual city. The only way to deal with this is to take inventory of regulations on a project-by-project basis.
- Specifications and descriptions of the (electronic) equipment that is part of the tested transmission channel: loudspeaker datasheets, device manuals, etc.
- Information on the test venue: blueprints, data on expected occupancy, expected background noise levels, placement of loudspeakers, etc.
- Interviews with the end customer and/or authority having jurisdiction (in case of systems used for evacuation purposes. Even a short phone call before the test plan is finalized may be enough to avoid unpleasant surprises further down the line, when the written test report is discussed.
- For complex projects, a visit to the test venue may be worth the invested time, just to get an impression of aspects that are not easily gleaned from documents and blueprints.

Information from this variety of sources has to be distilled into a test plan, which often takes the form of an itemized list:

- Which equipment to bring.
- Where to set the equipment up, and how to calibrate it.
- Which rooms/spaces to measure; how many measuring positions per room, and at which approximate locations; how many repeated measurements per location.
- Which post-processing actions to take.
- Other measurements and registrations besides STIPA, such as background noise level measurements.

Many standards require that all primary registrations of the measurements are logged. Sometimes test plans are written in a tabular format, that allows registration of measurement data right into the test plan document (which then also serves as a measurement log).

While the importance of having a solid test plan cannot be emphasized enough, it should also be noted that even the best plan usually has to be changed the very minute you enter the test venue. Even then, having a carefully considered and documented test plan helps; it will keep you on track while you try to be as flexible as possible in dealing with the unexpected.

4.2. Setting up equipment and calibrating

Before the actual start of measurements, all equipment has to be checked, charged, and calibrated. Next, the equipment has to be set up and prepared for the measurements.

- The talkbox has to be set up in front of the (paging) microphone, at the correct working distance and properly aligned. The sound level of the talkbox has to be verified against the reference speech level (usually 60 dB(A) measured at 1 meter distance).
- In scenarios where an electronic or file-based STIPA or full STI signal input is used, the signal level has to be matched to the level of spoken messages. This requires LAeq measurements of speech at a representative test location, followed by LAeq measurements for the STIPA signal. The STIPA signal is then adjusted to be 3 dB higher than the measured speech levels. Even better – the speech signal level can be tested without the Speech Level Meter module on the Bedrock SM50 or SM90, in which case no corrections are needed – simply adjust the STI signal to the same level as speech.
- Level calibration of the STI analyzer is checked by means of a level calibrator (if available) and adjusted if necessary.
- The STI analyzer is prepared for the measurements (e.g. mounted on a tripod adjusted to the correct measuring height).
- A so-called back-to-back measurement of the measuring system should *always* be done at the beginning of each test session. This takes just a few minutes, and will reveal any problems with your test equipment.

> *A back-to-back measurement is done by placing the STI analyzer close to the talkbox and performing STI and level measurements. Make sure that the talkbox and STA analyzer are positioned at least 1 meter away from reflecting surfaces, and place the tip of the microphone in front of the loudspeaker, at a distance of 0.25m. Now adjust the talkbox to 60 dB(A) at 1 meter, and carry out a STI measurement. Verify that the level is 72 dB(A) (± 1 dB), and that the STI>0.96.*

If the level or STI do not reach the target values, check for problems with your setup and make adjustments where needed. Clearly mark the back-to-back measurement in your log and save the measurement data on your STI analyzer.

Note that the Full STI and STIPA signal sound the same, but cannot be interchanged. Select the correct signal for the measurement module that you are using on your analyser.

<u>4.3. Walking measuring grids and collecting data</u>

Especially at large venues, the number of individual STI measurements may be quite high, depening on the application standard that you are following. In larger spaces, it helps to mark all measuring points on a map, or even physically by means of markers on the ground.

It is recommended to mount the STI analyzer on a tripod, for two reasons: it helps you to maintain the prescribed height (often 1.50m), and it allows for more stable measurements. It is good practice to take a step back from the device after starting the measurement, to minimize interference with the measurement.

While this phase comprises the bulk of the work to be done for any evaluation, the routine is usually quite easy. Problems with the transmission channel and test equipment have usually been caught in an earlier phase. Still, attention is needed to perform the measures correctly and stay alert for potential measurement issues:

- Stay clear from all reflective surfaces by at least 1 meter
- Hold the instrument away from your body, or step back from the tripod while measuring
- Continuously keep listening for interfering impulsive sounds (such as slamming doors) and signal interruptions. These events render the current measurement invalid, even if the STIPA analyzer does not generate a warning. If this happens, scratch the current measurement and start a new one.
- Keep monitoring whether results line up with the expected patterns, and keep checking if the STIPA analyzer is functioning normally without any warnings (e.g. battery level).

It is recommended to close off all measurement sessions with a final back-to-back measurement. This will allow you to rule out the possibility that problems with your test equipment started occurring halfway through your test session.

4.4. Reporting

Some application standards are very specific on the way data is to be reported. A common step inbetween the actual measurements and the generation of the final report is downloading your test data from the STIPA analyzer to a computer. After importing the data into a worksheet program such as Microsoft Excel, the data can be organized (e.g. by computing means and standard deviations) and processed further.

This is the phase where post-hoc calculations on the data are done. For instance, it is possible to computionally add noise to measurements done under noise-free conditions. More on this is given in section 7.

5. Specific requirements of NFPA72 (2013) Annex D

As an example of an application standard that uses STIPA tests, we give an overview of the specific requirements and details of NFPA72 (2013) [8] related to STIPA. NFPA Annex D gives guidelines for measuring speech intelligibility of Voice Evacuation systems with STIPA. This description does not fully cover all of the descriptions and requirements of NFPA72 Annex D, and cannot be used as a single source of information when you need to do measurements according to this standard. This section is just a summary of Annex D, focusing on the information directly related to STIPA testing.

In general, measurements for NFPA follow the same basic recipe as sketched in the preceding sections of this paper. However, there are a few additional requirements and a few deviations from the standard procedures.

5.1. Acoustically Distinguishable Spaces

NFPA72 uses the concept of the Acoustically Distinguishable Space (ADS). An ADS is a space (which can be an emergency zone by itself, or part of an emergency zone) that has clearly identifiable characteristics in terms of its acoustics. An ADS may be one room, or in larger rooms, it may be part of a room. If spaces are (nearly) identical in terms of background noise and room acoustics, then they are considered to be the same ADS and do not need to be measured twice. Needless to say that not having to measure in every single space can be a huge time saver in large projects.

5.2. Acceptability criteria

NFPA72 uses two acceptability criteria: the average STI in each ADS has to be at least 0.50, and at least 90% of the individual measurement points in an ADS must have an STI no less than 0.45 (section D.2.4.6). This leaves the option open to have up to 10% in "weak spots" within each ADS. These are criteria that are similar to most other (inter)national standards for evacuation systems.

If only one measurement is required in an ADS (see also below), then the STI at this location must be no less than 0.50. If a first measurement results in an STI<0.50, repeated measurements are taken, since some statistical spread in measurement results is to be expected. If the average is no less than 0.50, then the result is accepted.

5.3. Selection of measurement locations and test conditions

NFPA72 Annex D recognizes that intelligibility testing might not be necessary in every space. Reasons mentioned not to include a space for STIPA testing are the fact that a room is small (<30x30ft), the distance to speakers is small (<30ft), ceilings are low, and there are no hard surfaces such as marble or tile. Even in more challenging environments, NFPA72 allows for STIPA testing to be omitted, if the voice evac system is known to be designed by an experienced and skilled designer, making use of acoustic modeling software. In all other cases (larger rooms, hard surfaces, high ceilings), speech intelligibility testing is required.

For every ADS that requires intelligibility testing, measurement locations need to be selected. The basic rules are as follows:

- Measurement locations are separated by approx. 40ft. So in smaller rooms, there will be only one measurement point
- No more than 1/3 of the measurement points is allowed to be on the axis of a loudspeaker
- Points of particular interest should be included. Special attention goes to suspected worst-case locations (near noise sources) and egress paths.
- Three measurements are done at each location.

The measuring height is always 5ft (1.5 m). Measurement locations need to be mark on a plan (included as part of the test plan) for future reference.

To take the influence of background noise into account, NFPA72 allows occupied testing (when the background noise is representative), or unoccupied testing, in which case the occupied ambient noise is measured separately and included in results through post-hoc calculations. The latter option is less disruptive and therefore usually preferred; see section 7 of this section for a description of the procedures.

5.4. Setting up and calibrating

NFPA72 gives two options for calibrating the talkbox: a relatively complex procedure described in section D4 of the standard, or "according to the manufacturers instructions." We recommend the latter. For the Bedrock BTB65 TalkBox, this procedure is described in section 4.2 of this paper. Simply verify that the level at 0.25m distance is 12 dB higher than your reference speech level, and carry out a back-to-back STI measurement to confirm that the STI>0.96.

The procedure for matching the STIPA signal level to the level of speech (for electric signal input of file-based scenarios), the procedure is the same as the general procedure described in this paper. A wide tolerance margin is allowed; the levels of speech and STIPA signal are allowed to deviate up to ± 3 dB.

NFPA72 uses 65 dB(A) as a reference speech level instead of 60 dB(A). This is a remarkable and unorthodox choice. The rationale for using a higher vocal effort is absolutely valid: people usually have a tendency to speak louder when making voice announcements, in particular in an evacuation scenario. However, in the context of other standards, one might have expected 66 dB(A), as vocal effort is usually specified on a scale with 6 dB steps. This may be inconvenient with some talkboxes and signal generators, since default settings are usually based on these 6 dB steps.

> *For NFPA72, talkboxes need to be adjusted to different values than the normal reference value. The reference value according to NFPA72 is 65 dB(A). To investigate the influence of differences in vocal effort, the standard recommends doing measurements at 60 dB(A) and 70 dB(A) for at least one measurement location.*

5.5. Post-processing and reporting

All of the usual information is to be included in the test report; the standard gives an explicit itemized list:

- Building location and related descriptive facility information
- Names, titles, and contact information for individuals involved in test
- Dates and times of tests
- A list of testing instruments, including manufacturer's name, model, serial number, and date of most recent calibration
- Technical description of emergency communications system Identification of ADSs
- Locations of specific measurement points (in a list or on a set of drawings)
- Site definition of ambient sound pressure levels
- STI/STIPA measurements at each measurement point
- Final corrected STI/STIPA values where the post- processing procedure is used
- Indication of whether or not the test met the pass/fail criteria
- Record of system restoration (proving that the system is restored to its original state after the test)
- Any additional information to assist with future evaluation of system performance

The final result for each measurement point is calculated as the average of the three individual measurements at each point, corrected for ambient background noise (if applicable). For each ADS, the data from all measurement points combined results in a pass/fail conclusion.

6. Limitations of STIPA and Full STI

The Speech Transmission Index model, by its nature and design, has some limitations that render it less accurate (or even virtually useless) is specific situations. Whereas these situations may be rare in practice, it is always good to be aware of these limitations:

- Advanced digital low-bitrate voice coding systems do not accurately reproduce STI signals. STI signals are no real speech, but modulated noise; low-bitrate voice coders (such as systems used for military communications) actually suppress the STI signals instead of transmitting these. Cell phones can usually be tested, but not always – test results should be carefully scrutinized for inconsistency.
- Aggressive noise reduction algorithms also do not permit the use of noise-based test signals
- The STI is inaccurate for systems that feature center clipping. This was a common type of distortion in the days of carbon microphones, but it is very rare these days.

STIPA is an implementation of the STI that uses fewer modulation frequencies per octave band than the original "full STI". Due to a clever scheme that alternates modulation frequencies across octave bands, the entire modulation spectrum is still taken into account, but not for each octave band. This has been rigorously validated [] and proven to be accurate in almost every PA-related application (and virtually every other application as well). The only two exceptions are:

- whenever strong discrete echoes are observed, such as in some cathedrals and other very large venues, or when loudspeakers are spaced far apart without using delay lines to correct for the propagation time between one loudspeaker and the next.
- whenever strong reverberation (>3s) occurs with systems that feature bandwidth limiting (e.g. in which multiple octave bands are removed from the signal), such as lower quality PA systems.

The one limitation of both STIPA and full STI that requires constant attention is that the STI does not deal well with impulsive noises and signal interruptions.

> *In case of impulsive noises occurring during a STIPA measurement, such as door slams and accidental taps on the microphone, the STI measurement is rendered highly inaccurate. The same is true if the test signal has been interrupted during a measurement, however briefly. Some STIPA instruments (such as the Bedrock SM50) are capable of automatically detecting many, but not all, cases when measurements are invalidated. Even if the instrument does not generate a warning, discard a measurement whenever you observe impulse noises or signal interruptions.*

7. More advanced aspects of STI testing

7.1. Adding ambient noise through post-hoc calculations

The possibility to add in ambient noise through post-hoc calculations was mentioned several times before. This is one of many advanced post-processing options offered by the STI model. Most of these options are beyond the scope of this paper, but adding noise computationally is so common (and useful) that the procedure is described here.

If ambient noise is present during an STI measurement, its effect on intelligibility is taken into account automatically. Noise reduces the modulation depth of the STI signal, which is detected by the analyzer. But even if the noise is not physically present during the STI measurement, it can be included in the STI results, as long as the octave band spectrum is known. From the octave band spectrum, in combination with a "noise-free" STI measurement, the influence of the noise on the STI can be calculated after the fact. If the octave band noise spectrum has been measured accurately, this is just as reliable (and computationally equivalent) to measuring the STI directly in the presence of the ambient noise. The mathematics for this operation follow directly from the STI standard [5].

In practice, there are two alternative procedures to add ambient noise:

Procedure 1. On the STI analyzer	***Procedure 2. Afterwards in MS Excel***
• Measure the octave band spectrum of the ambient noise under representative conditions. Do NOT play the STI test signal during this measurement. • Some STI analyzers (such at the Bedrock SM50) allow you to enter the octave band noise spectrum directly into the STIPA or Full STI analyzer. • Now measure the STI under noise-free conditions. The STI analyzer provides two different STI numbers for each measurement: with and without (computationally added) noise.	• Measure the STI under noise-free conditions, covering all the data points of interest. Store the data. • Before or after the STI measurements, measure the octave band spectrum of the ambient noise under representative conditions. Do NOT play the STI test signal during this measurement. • After completion of all STI measurements, MS Excel tools (usually provided by the manufacturer of the STI analyzer) are used to add the noise to each STI measurement.

Which method is preferred depends on the situation. Procedure 1 is usually much more convenient, and allows you to evaluate the effect of the noise as you carry out your measurements. You will be able to adjust you test plan on the fly to address the impact of the ambient noise, if the need arises. However, not all STI analyzers give this option. Also, in same cases the ambient noise spectrum may not be known at the time the STI measurements take place. For instance, you may only have access to do elaborate STI tests before a venue is opened to the public, whereas you can only measure the ambient noise in the occupied state, after the venue has been opened to the public.

7.2. Level-dependent masking

Speech intelligibility partly depends on the absolute level of the speech. At very low speech levels as well as at high speech levels intelligibility decreases. This is due to imperfections of the human hearing organ, which suffers from a greater level of (internal) auditory masking at higher sound levels. This effect, known as level-dependent masking, is taken into account by the STI model.

However, in some cases, the actual level of the sound may not be known beforehand. For instance, evaluation of the performance of the combined electronics of a PA system is a task for which the STI is particularly suited. But these electronics may be used at a range of sound levels, from very high to very low. Incorporating the effect of level dependency does not make sense. For this reason, STIPA analyzers often give the option to disable level dependent masking. This option is sometimes also useful during back-to-back measurements: even if there is no signal distortion at all, at high sound levels (>75 dB(A)) the STI will never approach the theoretical maximum of 1.00. With level dependent masking disabled, STI-values >0.97 can be measured at any sound pressure level, as long as the transmission channel is completely free from interfering influences.

> *Disabling "level dependent masking" will lead to higher STI values at very low sound levels and at higher sound levels. Disabling level dependent masking is ONLY a valid choice when there is no acoustic reference, or when testing the measuring setup itself. Having level dependent masking disabled during any other kind of measurement leads to results that are too optimistic, and basically comes down to cheating.*

8. Conclusions and discussion

The Speech Transmission Index offers a convenient, relatively easy to use and reliable way to obtain objective data on speech intelligibility. This paper is an attempt to give an introductory overview of the procedures involved in STIPA and Full STI testing.

The Speech Transmission Index model opens up a wide spectrum of possibilities in terms of diagnostic analysis of speech transmission channels. This paper barely scratches the surface in that respect. Other resources are available that demonstrate how the STI helps troubleshoot systems, pinpointing root causes for intelligibility problems. Also, post-hoc calculations can be performed for much wider purposes that just adding background noise.

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About the authors of this paper

Sander van Wijngaarden obtained his Masters degree in applied physics from the Delft University of Technology in 1996. His specialization was Perceptual Acoustics. Sander then joined the Speech & Hearing group at TNO, where he worked on a broad range of R&D projects, mostly on hearing protection and speech intelligibility. He obtained his PhD from the Free University of Amsterdam in 2002. His PhD thesis ("The intelligibility of non-native speech") is based on work done at TNO.
After gaining experience as a project manager for several years, Sander was appointed team leader of the Speech & Hearing group in 2002. He took on a new challenge in 2006, when he joined a different division of TNO as a Business Developer. In 2007 Sander was appointed Department Manager in the Chemical and Biological protection department. Sander finally decided to leave TNO to co-found Embedded Acoustics in 2010, returning to his technological roots.

After obtaining his degree in electronics engineering in 1989, **Jan Verhave** joined the Phonetics department of the University of Amsterdam to work on a European "Esprit" project on speech technology. In 1993, he accepted a research associate position at TNO in Soesterberg. At TNO Jan specialized in real-time low-latency hard- and software for acoustic signal processing. Applications that Jan worked on include 3D audio, the Speech Transmission Index and Active Noise Reduction.
Jan was lead engineer and software developer in many development projects, including the first IEC-compliant STI measuring software application and several generations of Active Noise Reduction headsets (analog and digital). Jan was also the primary developer of STIPA and the novel Full STI method and was involved in several commercial implementations of STIPA. He has been the driving force behind most STI-related innovations over the last decade.

Currently, as founders and directors of Embedded Acoustics (the parent company of Bedrock Audio) Jan and Sander are still very actively involved in R&D and consulting in the field of speech intelligibility. They continue the work of their mentors (and STI inventors) Steeneken and Houtgast.