The Agreement Between a Transducer and a Microphone in the Analysis of a Synthesized Vowel—Using a Laboratory Model as a Preliminary Experiment

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SUMMARY: Objectives/hypothesis. The objective of this study was to assess the agreement between the acoustic parameters of synthesized vowels, as measured by a piezoelectric transducer and a microphone. The central hypothesis posited that there would be a substantial degree of concordance between the acoustic parameters obtained by these two methods.

Study design. A laboratory experiment was conducted using synthesized vowels. Acoustic parameters were recorded simultaneously using a microphone and a piezoelectric transducer.

Methods. Vowels were synthesized using *Madde* software, with variations in fundamental frequency (f_o) and amplitude level. Acoustic signals were captured by a Shure MX153 T microphone and a Shadow SH712 piezoelectric transducer. Signals were recorded using a Behringer FCA1616 audio interface and analyzed using *Sopran* and *VOXplot* software. Bland–Altman plots were used for statistical comparison of acoustic parameters.

Results. The degree of agreement between the microphone and the transducer varied according to acoustic parameters. Excellent agreement was observed for jitter, mean f_o, f_o standard deviation, and f_o range. Good agreement was found for tilt, equivalent continuous sound level, harmonics-to-noise ratio from Dejonckere, high frequency noise, minimum f_o, and maximum f_o. Moderate agreement was observed for slope, amplitude difference H1–H2, shimmer, harmonics-to-noise ratio, smoothed cepstral peak prominence, and period standard deviation.

Conclusions. While there was moderate to high agreement between the two methods for several acoustic parameters, discrepancies were noted. The microphone exhibited a tendency to record slightly higher values for the majority of the parameters; however, the transducer did demonstrate heightened sensitivity for certain parameters. The findings indicate that neither method can be universally interchanged with the other, and the selection of a method should be contingent on the particular requirements of the analysis.

Key Words: Piezoelectric transducer–Microphone–Synthesized vowels–Voice assessment–Acoustic parameters –Bland–Altman analysis.

INTRODUCTION

The microphone as a standard in voice capture and analysis

Microphones can capture a broad spectrum of frequencies, rendering them well-suited for the analysis of diverse voice characteristics. The selection of the appropriate microphone is crucial, considering factors such as frequency response, dynamic range, and directionality. A comprehensive understanding of the characteristics inherent in different microphone models is essential for ensuring the fidelity of the recorded signal.[1,2]

Factors affecting microphone reliability

Deliyski et al[3] conducted a study highlighting that microphone characteristics significantly influence voice quality measurements, stressing the need to take these characteristics into account in acoustic voice analysis methodologies. Additionally, Roy et al[4] underscored the significance of standardized procedures for microphone placement and recording, emphasizing how crucial such practices are to the reliability of the results obtained. Indeed, the reliability of results obtained from airborne microphones is contingent upon several factors.[1,3] These factors include the type of microphone utilized, the recording environment, calibration procedures, the analysis software, and the expertise of the researchers.[1,3,5,6] The microphone type affects the frequency response and directionality of the sound recorded.[5] Additionally, the recording environment can introduce noise and reverberation, which can impact measurement accuracy.[1] Calibration is essential for ensuring precise sound

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pressure level recordings.[6] Furthermore, the analysis software utilized and the expertise of the researchers can also influence the reliability of the results.[5]

Challenges in ideal signal acquisition

Capturing a signal that accurately represents the sound source requires adherence to stringent conditions.[6] However, these conditions are often challenging to meet, with failure to do so potentially compromising the quality and reliability of the acoustic signal. To mitigate this, recordings should ideally take place in acoustically controlled environments, such as sound-treated rooms.

The selection and positioning of the microphone are also of paramount importance. The use of a calibrated microphone in close proximity to the sound source has been shown to enhance the signal-to-noise ratio (SNR) and minimize variations in sound pressure levels.[6] Moreover, the utilization of calibrated recording equipment with appropriate settings ensures the accurate capture of sound pressure levels.[6] Failure to adhere to these standards can compromise the integrity of the acoustic data and the reliability of subsequent analyses.

Impact of ambient noise in acoustic voice analysis

Environmental noise constitutes a significant challenge, particularly for softer vocalizations.[3] Accurate measurement of ambient noise levels is crucial to ensure the reliability of acoustic voice analysis. Deliyski et al[3] documented the detrimental effects of uncontrolled ambient noise on voice quality measurements, noting that increased noise levels can lead to misinterpretation of results. The authors recommend maintaining a SNR of at least 30 dB and conducting recordings in sound-treated rooms to achieve this standard.

Their work validates earlier research, especially in the case of soft phonations,[5] and echoes previous recommendations for acoustically controlled environments, where ambient noise is minimized.[7,8] Additionally, the selection, placement, and calibration of microphones play a crucial role in ensuring accurate sound pressure level measurements.[1]

Different ways to record the human voice

Voice capture methods extend beyond traditional airborne microphones and can be categorized into two main types: contact-based and non-contact methods. Contact-based methods, which capture vibrations associated with voice production from the skin surface near the vocal folds,[9–11] entail the placement of sensors directly on the speaker's body to capture vibrations associated with voice production, including:

- *Neck-surface accelerometers:* Miniature accelerometers are attached to the speaker's neck, typically below the glottis, to measure vibrations of the tracheal wall. This method has shown promise in classifying phonation types and may offer advantages in noisy environments.[12,13] These sensors are also placed on the neck to capture vibrations related to subglottal pressure, providing insights into vocal function measures.[14]
- *Throat microphones:* These microphones are placed on the surface of the neck to capture vibrations from the vocal cords and surrounding tissues. The quality of the signal is contingent on the properties of the sensor and the manner in which vibrations propagate through the tissue and bone.[9–11]
- *Bone conduction microphones:* These microphones capture vibrations transmitted through the bones of the skull.[14–16]

Non-contact methods, such as traditional airborne microphones, employ a different approach, using sound waves in the air as the capture medium. However, these methods are prone to interference from background noise and distance from the speaker.[11,17–19] The selection of recording method is contingent upon the particulars of the application and the desired information. Contact-based methods have been shown to perform well in noisy environments, while non-contact methods are generally suitable for more general-purpose voice recording applications.

Potential of contact-based methods in acoustic analysis

Contact-based methods have been demonstrated to effectively mimic the function of microphones in acoustic voice analysis, particularly in situations where traditional microphones are impractical.[11–13] However, the accuracy of these methods can be influenced by sensor properties and the propagation of vibrations through tissue and bone. Despite these challenges, contact-based methods offer several advantages, including reduced susceptibility to ambient noise, consistent signal quality regardless of head movements, and less obtrusiveness during recording.[11,12]

Hauret et al[20] have built upon the foundations laid by previous research on body-conducted speech analysis by introducing a novel dataset, Vibravox, which comprises recordings from five distinct body-conduction microphones and an airborne microphone. This study makes a significant contribution to the field by offering a more nuanced understanding of the potential and challenges of body-conduction microphones in the context of speech capture and processing. In another study, Cantor-Cutiva et al[13] investigated the agreement between microphone and neck-worn accelerometer recordings in voice acoustic parameters. Their findings offer further substantiation for the reliability of specific voice parameters across diverse recording methodologies and contribute to a more profound comprehension of the elements that affect voice acoustic measurements.

Piezoelectric transducers as an alternative recording method

In light of the difficulties in attaining optimal recording conditions, particularly in noisy environments, this article proposes the use of piezoelectric transducers as an alternative to conventional microphones in the capture of vocal signals. A piezoelectric transducer placed on the skin of the neck at larynx level facilitates the conversion of mechanical vibrations into electrical signals.

The operational principle of piezoelectric transducers relies on the piezoelectric effect, where certain materials generate an electric charge in response to mechanical stress.[21–23] These transducers have applications in various fields, including sensors and energy harvesting systems.[24,25]

In the context of voice analysis, piezoelectric transducers are used in throat microphones, which utilize the principle to capture vibrations from the vocal folds.[9,10] The quality of the signal is contingent upon the sensor's properties, the propagation of vibrations through the skin and bone, and the coupling between the sensor and the skin.

Piezoelectric transducers offer several advantages for voice analysis, including reduced noise interference, improved signal quality, and increased comfort and mobility.[26] However, they also have limitations, such as sensitivity to temperature changes and potential interference from other body vibrations.[27]

Study aim

This study aims to assess the level of agreement between the acoustic parameters of a piezoelectric transducer and a microphone in analyzing synthesized vowels. The proposed use of piezoelectric transducers seeks to address the limitations of traditional airborne microphones in capturing clear vocal signals in challenging acoustic conditions.

The central hypothesis of this study is that there is a high degree of agreement between the acoustic parameters of a piezoelectric transducer and a microphone in the analysis of synthesized vowels.

It is crucial to emphasize that this investigation utilizes a laboratory model with synthesized vowels and a loudspeaker to provide a controlled and replicable acoustic source. The primary aim is to understand the fundamental agreement and inherent differences between these two sensor types when responding to identical vibratory events under idealized conditions. This approach, while not replicating the complexities of human voice production and tissue propagation, serves as a necessary preliminary step to isolate sensor-specific characteristics before extending investigations to human participants, where such factors would introduce significant additional variables. The findings are therefore intended to provide foundational knowledge about the sensors themselves rather than to be directly extrapolated to *in vivo* voice assessment.

METHODS

Experiment setup

The choice of a loudspeaker as the sound/vibration source, while acknowledging its distinct differences from the human vocal production mechanism, was predicated on the need for a highly controllable and replicable stimulus emitter in this preliminary laboratory investigation. The loudspeaker cone provides a consistent vibrating surface, allowing for a foundational comparison of the transduction properties of the contact-based piezoelectric sensor versus the airborne microphone to identical synthesized acoustic events. This approach aims to understand the baseline sensor agreement before addressing the complexities of *in vivo* application, such as propagation through and vibration of cervical tissues.

Recognizing the critical importance of a favorable SNR for accurate acoustic measurements, as emphasized by Švec and Granqvist,[6] efforts were made to optimize this condition. Analysis of silent portions of the recordings using *Sopran* (version 1.0.28, Svante Granqvist, Tolvan Data, Stockholm, Sweden) indicated a baseline noise level of approximately 39.8 dBC, suggesting an adequate SNR for the relatively high-intensity synthesized vowel signals used. Also, perceptual monitoring of possible noises was performed.

Synthesis of vowels

Vowel synthesis was performed using the *Madde* software (version 3.0.0.2, Svante Granqvist, Tolvan Data). Generation of six distinct vowel samples was achieved through the manipulation of key acoustic parameters, thereby reflecting varying voice characteristics while maintaining all other software settings at their default values. The specific parameters to undergo modification were the fundamental frequency (f_o) and the amplitude level. The f_o was set at 3 different values—110 Hz, 220 Hz, and 440 Hz—representing a typical male voice frequency, a typical female voice frequency (1 octave higher), and a value precisely 1 octave above the female frequency setting.[28,29] For each frequency, 2 amplitude levels were selected: −30 dB, corresponding to a typical modal loudness, and −40 dB, representing a softer vocal intensity. This systematic approach aimed to ensure a consistent and controlled variation of acoustic parameters, facilitating the subsequent analysis of their effects on the study's outcomes.

Sound playback and signal amplification

The audio playback and the amplification system were set up to ensure accurate and reliable reproduction of the synthesized vowel sounds. The setup included the following components:

- *Loudspeaker*: The M-Audio BX5 (inMusic Brands, Cumberland, Rhode Island, USA) is a monitor-grade speaker that was utilized for the purpose of sound playback.
- *Audio interface*: The connection between the playback software and the loudspeaker was facilitated by a Focusrite Scarlett 2i2 audio interface (Focusrite PLC, High Wycombe, United Kingdom), which provided digital-to-analog conversion.
- Sound playback software: *Sopran* (version 1.0.28) was used to play the synthesized vowels.
- Computer: The playback setup was controlled by a Lenovo Thinkpad 6th Gen computer (Lenovo Group Limited, Morrisville, North Carolina, USA) running Windows 11 (Microsoft Corporation, Redmond, Washington, USA). This computer was used to run the *Sopran* software and manage the audio signals routed through the interface to the loudspeaker.

Recording equipment and setup

To capture both airborne and contact-based audio signals, a dual recording setup was implemented (Figure 1). This approach ensured simultaneous recording of the microphone and the piezoelectric transducer, using the same audio interface to maintain synchronization and consistency across all recordings. This section was designed to align with the most recent guidelines from Švec and Granqvist.[1]

- *Microphone*: A Shure MX153 T earset microphone (Shure Incorporated, Niles, Illinois, USA) with an omnidirectional polar pattern was used to capture the airborne sound. The microphone was attached to the loudspeaker and the capsule was positioned 5 cm away from the center of the loudspeaker, in proximity to both the woofer and the tweeter, though primarily aligned with the woofer's acoustic axis.
- *Piezoelectric transducer*: A Shadow SH712 (Shadow Electronics GmbH, Germany) dual-element piezoelectric contact transducer was used to capture vibrations directly from the sound source. The transducer was attached to the membrane of the loudspeaker's woofer. To ensure optimal coupling and prevent mechanical noise or "rattling," a thin layer of reusable adhesive putty was applied directly to the contact surface of the transducer, which was then pressed firmly onto the woofer cone. Mefix tape (Mölnlycke Health Care, Gothenburg, Sweden) was subsequently used over the transducer to secure its position consistently throughout the experiment. The integrity of the contact and the absence of extraneous noise were verified by perceptual monitoring of the recorded signals.
- *Audio interface*: A Behringer FCA1616 audio interface (Music Tribe, Makati, Philippines) was used to capture the signals from both the microphone and the transducer. This interface facilitated high-resolution analog-to-digital conversion with a sampling rate of 96 kHz and a bit depth of 24 bits, low-latency performance, and multiple input options. The microphone was connected to an XLR (External Line Return) input of the interface. The piezoelectric transducer was connected to a dedicated high-impedance (Hi-Z) 1/4 in. TRS (Tip-Ring-Sleeve) instrument input on the audio interface, ensuring appropriate impedance matching for optimal signal capture from the passive transducer. The potentiometers of the built-in preamplifier were configured to enable maximum input, thereby ensuring that signal clipping was not occurring.
- *Computer*: A mid-2013 MacBook Pro running macOS Mojave (10.14.6, Apple Inc, Cupertino, California, USA) was used for recording and storing the audio data. The audio interface was connected to the computer via a FireWire 800 cable (Lindy-Elektronik GmbH, Mannheim, Germany).
- *Software*: *Logic Pro X* (version 10.4.1, Apple Inc) digital audio workstation software was used for recording and exporting audio data. The software was used to simultaneously capture the two synchronized signals from the microphone and piezoelectric transducer, facilitating accurate temporal alignment of the recordings. Each signal was recorded on a separate track within the software. Following recording at 96 kHz/24 bits, the audio data were exported from *Logic Pro X* as separate WAV files with a bit depth of 16 bits and a sample rate of 44.1 kHz, utilizing the software's inherent resampling algorithms. This sample rate was chosen for compatibility with the analysis software used. It should be noted that no plugins or additional signal processing, beyond the aforementioned resampling during export, were applied to the recorded audio within *Logic Pro X*. This approach aimed to preserve the core characteristics of the signals from the microphone and transducer for subsequent analysis.

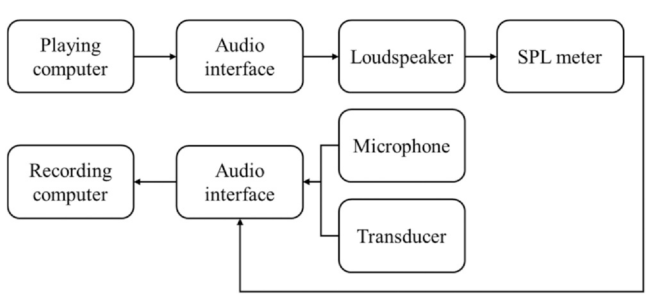

FIGURE 1. Experimental setup for acoustic data acquisition.

Calibration and acoustic measurements

This section was structured to align with the most recent guidelines from Švec and Granqvist.[6]

- *Sound pressure level measure*: A sound level meter model AZ 8922 (AZ Instrument Corp., Taichung, Taiwan) was used to assess the acoustic pressure levels. The distance from the meter to the loudspeaker was maintained at 30 cm, ensuring consistent height throughout the experiment. The meter was configured with C-weighting.
- *Calibration*: *Sopran* (v.1.0.28, Svante Granqvist) was employed for the analysis and manipulation of audio signals. Audio files of the vowels, recorded from both the microphone and the transducer, were opened and combined into a single *Sopran* software native file format for the purpose of facilitating analysis. The initial vowel from the microphone recording was played, and the sound level meter reading was observed. The resultant value of 78.4 dB was documented, thereby establishing a reference point for calibration in *Sopran*. The same section of the first vowel was selected in both the microphone and transducer recordings, thus enabling direct comparison and adjustment of the levels between the two signals, ensuring complete alignment.
- *Audio file creation*: Extraction of individual vowel sounds from the recordings was conducted, with the resultant files being saved as separate WAV files for

TABLE 1.
Acoustic Parameters Extracted for Comparative Analysis

Abbreviation (Unit of Measurement)	Description
Leq (dBC)	Equivalent continuous sound level (C-weighted)
Amplitude difference H1–H2 (dB)	Amplitude difference between H1 and H2
CPPS	Cepstral peak prominence smoothed
Mean f_o (Hz)	Average fundamental frequency
f_o Standard deviation (Hz)	Fundamental frequency standard deviation
Minimum f_o (Hz)	Minimum fundamental frequency
Maximum f_o (Hz)	Maximum fundamental frequency
f_o Range (semitones)	Fundamental frequency range
Slope (dB)	Global energy distribution across the power spectrum
Tilt (dB)	The slope of the regression line through the Long-Term Average Spectrum (LTAS)
Jitter (%)	The fundamental frequency perturbation
Shimmer (%)	The amplitude perturbation
PSD (ms)	Period standard deviation
HNR (dB)	The ratio of harmonics to noise components in the excitation signal. HNR stands for Harmonics-to-Noise Ratio
HNR-D (dB)	The harmonic structure in the frequency range between 500 Hz and 1500 Hz. HNR-D stands for Harmonics-to-Noise Ratio from Dejonckere
GNE	Glottal-to-noise-excitation ratio
HFN (dB)	High frequency noise

the microphone and transducer signals. This approach enables individual analysis of each vowel.

- *Acoustic parameters measurements*: Two software programs were used to extract the parameters presented in Table 1: *Sopran* (1.0.28, Svante Granqvist) and *VOXplot* (v.2.1.0, Jörg Mayer, Lingphon, Straubenhardt, Germany).[30]
- *Data management*: A comprehensive datasheet was utilized for the systematic storage of all parameters.

Comparison of methods

Bland–Altman plots. This statistical method was employed to compare the measurements obtained from the microphone and the piezoelectric transducer. Bland–Altman plots are employed to graphically represent the agreement between the two quantitative measurement techniques, thereby identifying any systematic biases or inconsistencies.[31] The generation of these plots and the underlying data were facilitated by *MedCalc* (v.14.8.1, MedCalc Software Ltd, Ostend, Belgium). While ideally, acceptable limits for agreement should be defined a priori based on clinical or practical significance,[32] such benchmarks are not established for this specific novel comparison of a microphone and a piezoelectric transducer analyzing synthesized vowels from a loudspeaker. Therefore, the interpretation of agreement levels in this exploratory study (Table 4) was based on a descriptive, post hoc categorization (Excellent, Good, Moderate) considering the magnitude of the observed mean difference (bias) and the width of the 95% limits of agreement relative to the typical range and variability of each acoustic parameter as reported in voice research literature. This approach provides a qualitative assessment of the degree of concordance.

RESULTS

Table 2 presents a comparison of the mean difference (bias), the limits of agreement and the amplitude between the microphone and the transducer for each parameter. The mean difference for most parameters is small, and the limits of agreement are relatively narrow. The parameters with the greatest discrepancies between the microphone and the transducer are the amplitude difference H1–H2 and Shimmer. The agreement between the microphone and the transducer are depicted in the Bland–Altman plots for each acoustic parameter (Figures 2–18). The detailed statistics for each Bland–Altman plot are shown in Table 3. A comprehensive assessment of the outcomes pertaining to each acoustic parameter is provided in Table 4, with specific focus on the mean difference (bias) and the limits of agreement.

Table 4 presents an interpretive classification of the limits of agreement. Given the absence of a universally accepted standard for the classification of acoustic voice analysis, this study has developed a set of criteria based on the magnitude of the limits of agreement in relation to the typical range of values for each parameter.

Overall, the results of the Bland–Altman analysis suggest that there is a high degree of agreement between the microphone and the transducer for most acoustic parameters.

Bias

The microphone generally exhibits a tendency to record slightly higher values for most of the measurements, except

TABLE 2.
Quantitative Comparison of Acoustic Parameters Measured by Microphone and Transducer

Parameter	Mean Difference (Bias)	Limits of Agreement	Amplitude
Leq (dBC)	0.9	−2.9 to 4.6	7.5
Amplitude difference H1–H2 (dB)	−3	−16.9 to 10.9	27.8
CPPS	0.8	−1.6 to 3.3	4.9
Mean f_o (Hz)	0	−0.05 to 0.05	0.1
f_o Standard deviation (Hz)	−0.02	−0.07 to 0.02	0.09
Minimum f_o (Hz)	−0.36	−1.57 to 0.84	2.41
Maximum f_o (Hz)	0.32	−0.95 to 1.60	2.55
f_o Range (semitones)	0.02	−0.09 to 0.13	0.22
Slope (dB)	1.6	−4.1 to 7.3	11.4
Tilt (dB)	−1.8	−4.8 to 1.2	6
Jitter (%)	0	−0.07 to 0.07	0.15
Shimmer (%)	−1.1	−8.0 to 5.9	13.9
PSD (ms)	−0.09	−0.70 to 0.52	1.22
HNR (dB)	1.4	−8.6 to 11.4	20
HNR-D (dB)	1.4	−2.1 to 4.8	6.9
GNE	−0.007	−0.1 to 0.04	0.0902
HFN (dB)	1.08	−0.23 to 2.39	2.62

Abbreviations: CPPS, cepstral peak prominence smoothed; GNE, glottal-to-noise-excitation; HFN, high frequency noise.

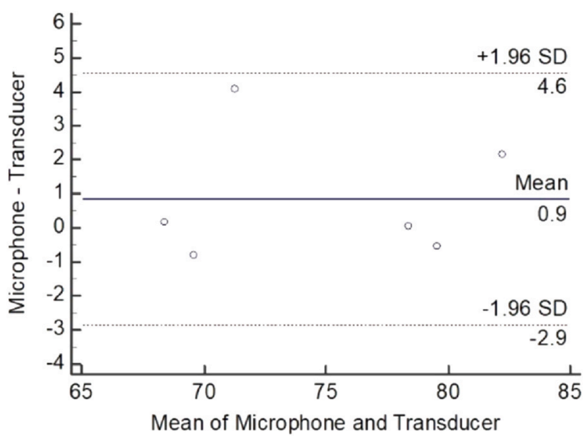

FIGURE 2. Bland–Altman plot for the Leq (dBC).

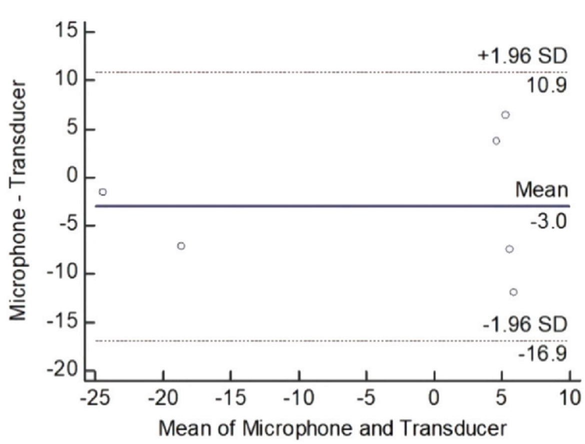

FIGURE 3. Bland–Altman plot for the H1–H2 (dB).

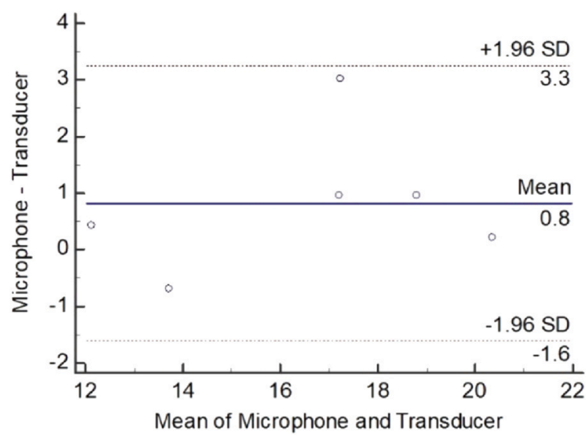

FIGURE 4. Bland–Altman plot for the CPPS. CPPS, cepstral peak prominence smoothed.

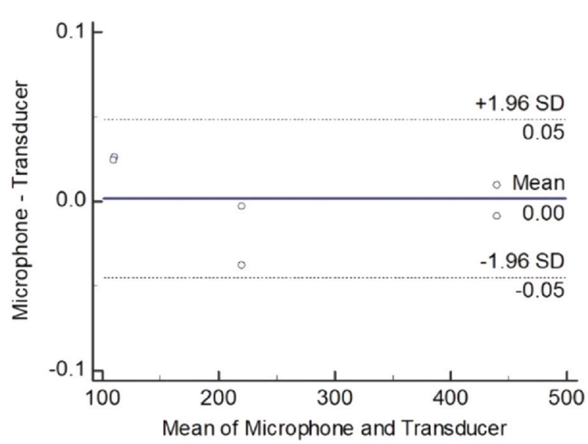

FIGURE 5. Bland–Altman plot for the mean f_o (Hz).

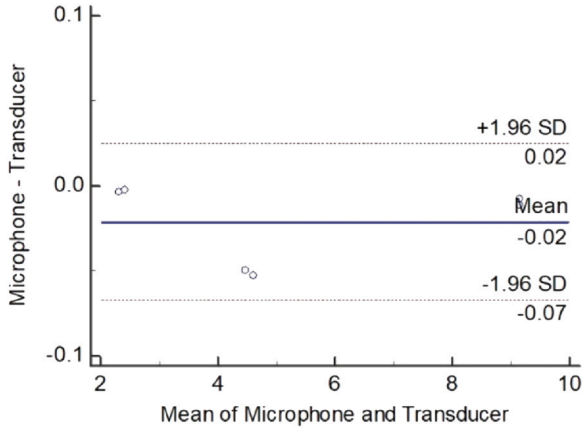

FIGURE 6. Bland–Altman plot for the f_o standard deviation (Hz).

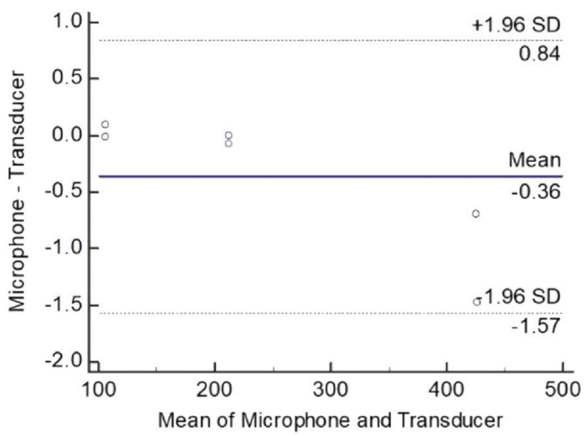

FIGURE 7. Bland–Altman plot for the minimum f_o (Hz).

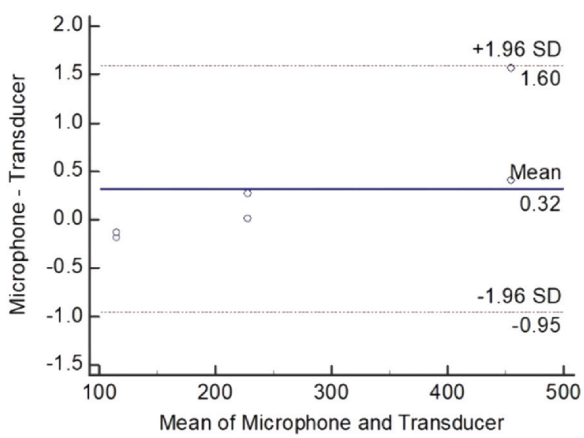

FIGURE 8. Bland–Altman plot for the maximum f_o (Hz).

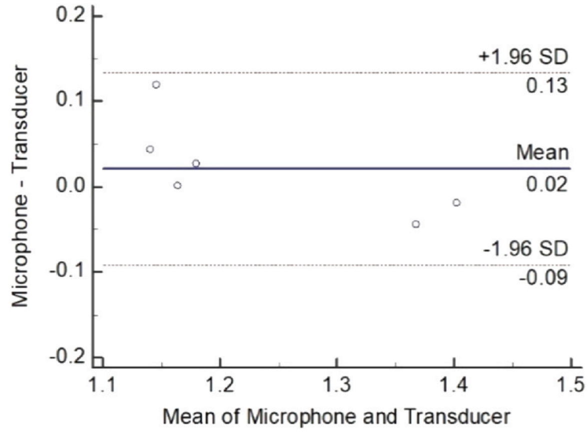

FIGURE 9. Bland–Altman plot for the f_o range (semitones).

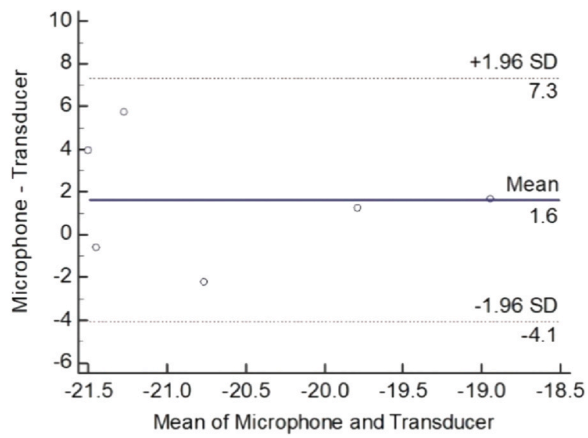

FIGURE 10. Bland–Altman plot for the slope (dB).

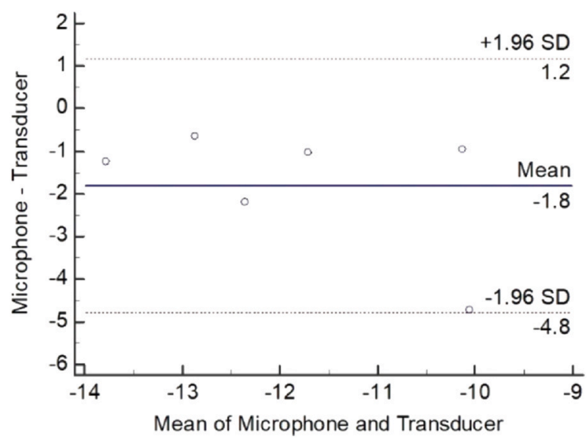

FIGURE 11. Bland–Altman plot for the tilt (dB).

for tilt, amplitude difference H1–H2, shimmer, period standard deviation (PSD), and minimum f_o, where the transducer demonstrates higher values. The variability between the two methods is generally moderate, with certain parameters exhibiting greater variability than others. Some graphs reveal the potential presence of outliers, indicating that agreement can vary significantly between individuals.

While the two methods generally demonstrate adequate agreement for most parameters, subtle discrepancies exist that must be considered during the analysis and interpretation of results.

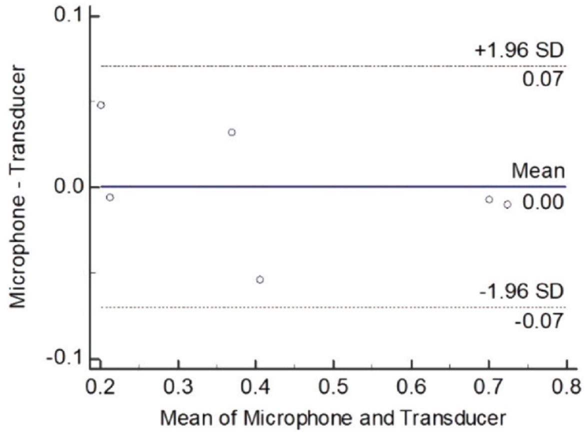

FIGURE 12. Bland–Altman plot for the jitter (%).

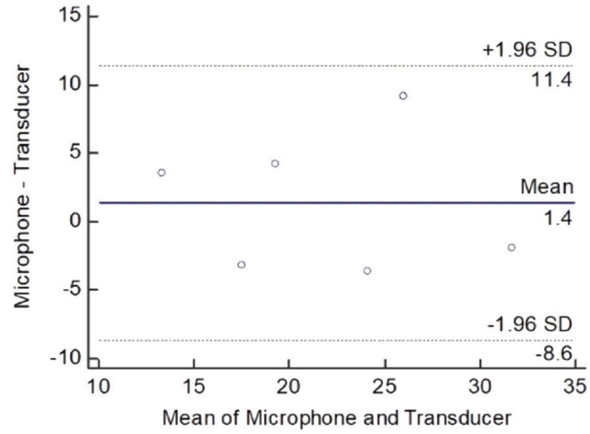

FIGURE 15. Bland–Altman plot for the HNR (dB).

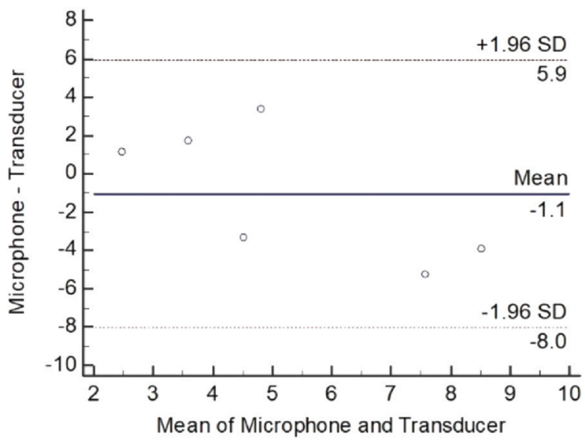

FIGURE 13. Bland–Altman plot for the shimmer (%).

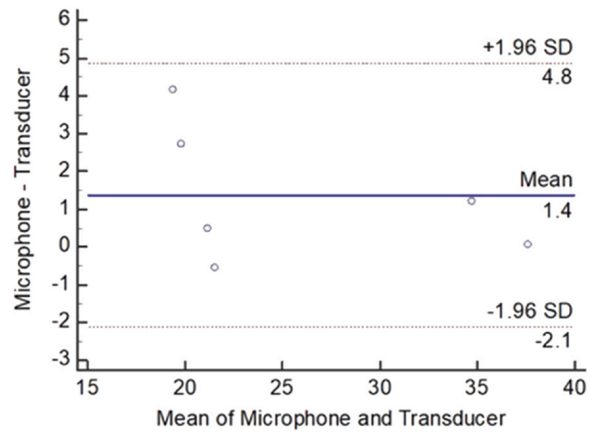

FIGURE 16. Bland–Altman plot for the HNR-D (dB).

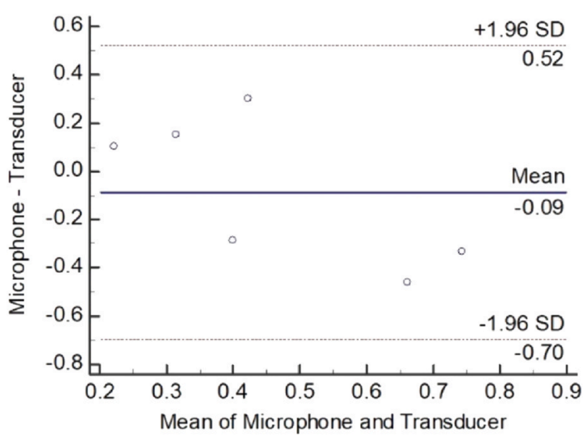

FIGURE 14. Bland–Altman plot for the PSD (ms).

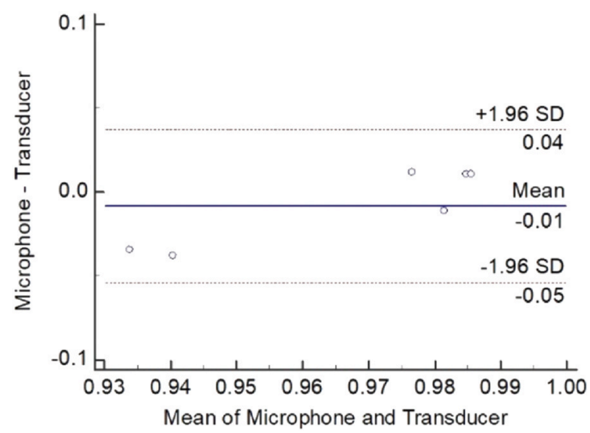

FIGURE 17. Bland–Altman plot for the GNE. GNE, glottal-to-noise-excitation.

Limits of agreement

Excellent agreement was found for jitter, mean f_o, f_o standard deviation, and f_o range, with minimal bias and narrow limits of agreement. This finding indicates that both methods produce very similar results for these measurements. Good agreement was found for tilt, Leq, HNR-D, high frequency noise, minimum f_o, and maximum f_o. Despite a slight tendency towards one of the methods, the limits of agreement were relatively narrow, indicating that the observed differences between the measurements are generally minimal. Moderate agreement was observed for slope, amplitude difference H1–H2, shimmer, HNR, cepstral peak prominence smoothed, and PSD. In these cases,

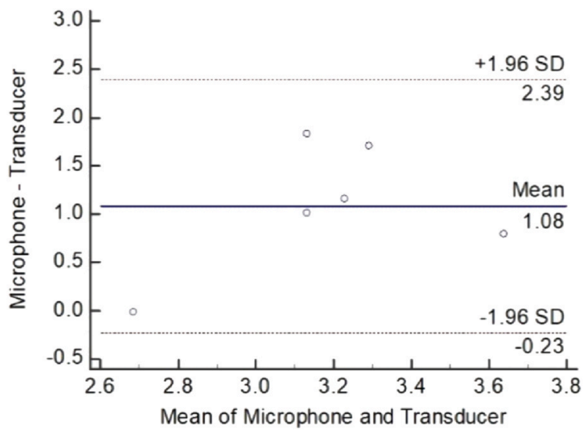

FIGURE 18. Bland–Altman plot for the HFN (dB). HFN, high frequency noise.

although the bias was generally small, the variability was greater, indicating that the differences between the measurements may be more significant in some cases.

DISCUSSION

This study sought to ascertain the degree of concordance between measurements of acoustic parameters in synthesized vowels obtained using a microphone and a piezoelectric transducer. The findings indicated a range of levels of agreement, underscoring the intricacies inherent in the capture and analysis of acoustic signals.

Methodological innovation often involves combining established principles in novel configurations to address specific research questions. While direct precedents for the exact experimental configuration used here may not exist, the individual components and the underlying logic of creating a controlled environment for sensor comparison are scientifically grounded.

Furthermore, the classification of agreement levels (Excellent, Good, Moderate) was based on a post hoc descriptive approach relative to the parameters' typical ranges, rather than on a priori defined limits of acceptable difference based on practical or clinical significance, as such limits are not established for this specific experimental context. This should be considered when interpreting the potential interchangeability of the methods.

The observed tendency for the microphone to produce lower limits, Leq suggests the possibility of differential sensitivity between the two methods. This discrepancy can be ascribed to the distinct mechanisms by which each device captures sound: the microphone measures airborne sound pressure, whereas the transducer measures mechanical vibrations. The transducer's contact-based nature and attachment to the loudspeaker may influence its sensitivity to the overall sound energy compared to the microphone's free-field measurement. Furthermore, potential variations in the coupling of the transducer to the sound source may also contribute to the observed differences.

The bias towards the transducer in measuring the amplitude difference H1–H2 is particularly interesting, suggesting that the transducer may be more sensitive to variations in the harmonic structure of the signal. The observed discrepancy may be attributable to variations in the frequency response of the two devices or to the transducer's capacity to detect vibrations in closer proximity to the source. This capability could potentially diminish the impact of resonance effects that are typically captured by the microphone. Further exploration, employing more sophisticated methodologies such as modal analysis, might provide a more comprehensive understanding of the underlying causes of these variations.

The minimal bias observed for mean f_o aligns with the expectation that f_o, a primary acoustic feature, would be captured consistently by both methods. This consistency supports the reliability of both devices for measuring this parameter. However, the discrepancies observed for other parameters, such as shimmer and HNR, indicate that while both methods capture the essence of the acoustic signal, they may represent finer details differently.

These findings must be interpreted with considerable caution, primarily within the context of the study's significant limitations as a preliminary laboratory model. The utilization of synthesized vowels produced by a loudspeaker, while providing a highly controlled and replicable acoustic environment for comparing the intrinsic responses of the two sensor types, fundamentally differs from the complexities of human voice production. Crucially, this model does not incorporate the biomechanical influences of sound and vibration propagation through laryngeal structures, the vocal tract, and cervical tissues, nor does it account for the variable coupling of a transducer to the skin surface. These factors are expected to be major determinants of the signals captured in *in vivo* recordings and would likely lead to different agreement patterns than those observed here. Therefore, the current results primarily shed light on the inherent transduction characteristics of the microphone versus the piezoelectric transducer when presented with a common, simplified acoustic/vibratory event, and cannot be directly extrapolated to predict their agreement in human voice assessment. The specific placement and attachment of the transducer to the loudspeaker woofer, although standardized within this study, also represent a condition distinct from application on human skin. This study should thus be viewed as a foundational step, exploring sensor behavior in a controlled setting, antecedent to more ecologically valid research.

In comparison with the study by Cantor-Cutiva et al,[13] which concentrated on the agreement between microphone and accelerometer recordings of human speech, this study offers a complementary perspective by examining the use of a piezoelectric transducer with synthesized vowels. While Cantor-Cutiva et al[13] highlighted the influence of recording timing and speech type, this study contributes by drawing attention to the nuances in capturing different acoustic parameters with contact-based and airborne methods. Both studies underscore the importance of careful consideration when selecting and interpreting data from different recording devices.

TABLE 3.
Descriptive Statistics for Bland–Altman Analysis of Acoustic Parameters

Parameter	Sample Size	Arithmetic Mean	95% Confidence Interval (CI)	*P* (H_0: Mean = 0)	Standard Deviation (SD)	Lower Limit	95% Confidence Interval	Upper Limit	95% Confidence Interval
Leq (dBC)	6	0.86	−1.13 to 2.84	0.32	1.89	−2.85	−6.46 to 0.76	4.56	0.95 to 8.17
Amplitude difference H1–H2 (dB)	6	−2.99	−10.42 to 4.45	0.35	7.08	−16.88	−30.39 to −3.36	10.9	−2.62 to 24.41
CPPS	6	0.82	−0.48 to 2.12	0.17	1.24	−1.61	−3.97 to 0.76	3.25	0.89 to 5.62
Mean f_o (Hz)	6	0.002	−0.02 to 0.03	0.87	0.02	−0.05	−0.1 to 0.001	0.05	0.003 to 0.09
f_o Standard deviation (Hz)	6	−0.02	−0.05 to 0.003	0.08	0.02	−0.07	−0.11 to −0.02	0.02	−0.02 to 0.07
Minimum f_o (Hz)	6	−0.36	−1.01 to 0.28	0.21	0.61	−1.57	−2.74 to −0.4	0.84	−0.33 to 2.02
Maximum f_o (Hz)	6	0.32	−0.36 to 1.01	0.28	0.65	−0.95	−2.2 to 0.29	1.6	0.36 to 2.84
f_o Range (semitones)	6	0.02	−0.04 to 0.08	0.4	0.06	−0.09	−0.2 to 0.02	0.13	0.02 to 0.24
Slope (dB)	6	1.63	−1.42 to 4.69	0.23	2.91	−4.07	−9.62 to 1.48	7.33	1.78 to 12.89
Tilt (dB)	6	−1.80	−3.40 to −0.20	0.03	1.52	−4.79	−7.69 to −1.88	1.18	−1.72 to 4.09
Jitter (%)	6	0.0005	−0.04 to 0.04	0.97	0.03	−0.07	−0.14 to −0.001	0.07	0.002 to 0.14
Shimmer (%)	6	−1.06	−4.79 to 2.68	0.5	3.56	−8.03	−14.82 to −1.24	5.92	−0.87 to 12.71
PSD (ms)	6	−0.0006	−0.001 to −0.00008	0.03	0.0005	−0.002	−0.002 to −0.001	0.001	−0.0006 to 0.001
HNR (dB)	6	1.3608	−3.1 to 6.72	0.54	5.1	−8.64	−18.38 to 1.09	11.37	1.63 to 21.11
HNR-D (dB)	6	1.36	−0.51 to 3.22	0.12	1.78	−2.13	−5.53 to 1.26	4.84	1.45 to 8.24
GNE	6	−0.008	−0.03 to 0.02	0.41	0.02	−0.05	−0.1 to −0.01	0.04	−0.007 to 0.08
HFN (dB)	6	1.08	0.38 to 1.78	0.01	0.67	−0.23	−1.5 to 1.05	2.39	1.12 to 3.66

Abbreviations: CPPS, cepstral peak prominence smoothed; GNE, glottal-to-noise-excitation; HFN, high frequency noise.

TABLE 4.
Interpretation of Agreement Between Microphone and Transducer Measurements

Parameter	Mean Difference (Bias)	Limits of Agreement
Leq (dBC)	Slight bias towards microphone	The limits of agreement are moderately wide, indicating some variability in the Leq measurements between the microphone and transducer.
Amplitude difference H1–H2 (dB)	Bias towards transducer	The limits of agreement are quite wide, suggesting substantial variability in the H1–H2.
CPPS	Bias towards microphone	The limits of agreement are moderately wide, indicating some variability in CPPS measurements.
Fundamental frequency		
Mean f_o (Hz)	Minimal bias	The limits of agreement are extremely narrow, indicating excellent agreement between the microphone and transducer for measuring mean frequency.
f_o Standard deviation (Hz)	Minimal bias	The limits of agreement are very narrow, indicating excellent agreement between the microphone and transducer for measuring the standard deviation of frequency.
Minimum f_o (Hz)	Bias towards transducer	The limits of agreement are moderately wide, indicating some variability in minimum frequency measurements between the microphone and transducer.
Maximum f_o (Hz)	Bias towards microphone	The limits of agreement are moderately wide, indicating some variability in maximum frequency measurements between the microphone and transducer.
f_o Range (semitones)	Minimal bias	The limits of agreement are very narrow, indicating excellent agreement between the microphone and transducer for measuring vocal range in semitones.
Spectral slope		
Slope (dB)	Slight bias towards microphone	The limits of agreement are moderately wide, indicating some variability in the spectral slope measurements between the microphone and the transducer.
Tilt (dB)	Bias towards transducer	The limits of agreement are relatively narrow, suggesting good agreement between the microphone and transducer for measuring spectral tilt.
Perturbation		
Jitter (%)	Minimal bias	The limits of agreement are very narrow, indicating excellent agreement between the microphone and transducer for measuring local jitter.
Shimmer (%)	Bias towards transducer	The limits of agreement are relatively wide, indicating a considerable degree of variability in local shimmer measurements between the microphone and the transducer.
PSD (ms)	Slight bias towards transducer	The limits of agreement are relatively narrow, suggesting good agreement between the microphone and transducer for measuring PSD.
Harmonicity		
HNR (dB)	Slight bias towards microphone	The limits of agreement are quite wide, indicating substantial variability in HNR measurements between the microphone and transducer.
HNR-D (dB)	Slight bias towards microphone	The limits of agreement are moderately wide, suggesting some variability in HNR-D measurements between the microphone and transducer.
GNE	Minimal bias	The limits of agreement are extremely narrow, suggesting that for most individuals, the difference in GNE measured by the microphone and transducer would be very small.
HFN (dB)	Bias towards microphone	The limits of agreement are relatively narrow, suggesting good agreement between the microphone and transducer for measuring HFN.

Notes: Bias towards microphone: The mean difference indicates that the microphone tends to record slightly higher parameter values than the transducer.
Bias towards transducer: The mean difference indicates that the transducer tends to record higher parameter values than the microphone.
Minimal bias: The mean difference indicates negligible or virtually no systematic difference between the microphone and transducer in measuring parameter values.
Abbreviations: CPPS, Cepstral peak prominence smoothed; GNE, glottal-to-noise-excitation; HFN, high frequency noise.

A more comprehensive interpretation of the results of this study can be achieved by considering the findings of previous research that has examined contact-based methods for voice analysis.[9–11,15,19,33–37] Studies have explored the use of neck-surface accelerometers, throat microphones, and bone conduction microphones for capturing voice signals.[9–12,15,19,36,37]

This study utilized a piezoelectric transducer positioned on the skin of the neck, analogous to a throat microphone. Such devices have demonstrated potential for voice analysis, with the quality of the signal being contingent on factors such as sensor properties and vibration propagation through tissue. Švec et al[9] have demonstrated the feasibility of estimating sound pressure levels from skin vibration using an accelerometer on the neck.

Research has also been conducted comparing contact-based methods to traditional microphones. A study by Coleman[36] compared microphone and accelerometer monitoring of the performing voice, finding differences in amplitude characteristics.

Furthermore, the loudspeaker used is a two-way system. While the transducer was coupled to the woofer, the microphone captured sound from both the woofer and the tweeter. Although the microphone was positioned in close proximity to the woofer, this difference in capturing the full spectrum radiated by the loudspeaker system might have contributed to some discrepancies, particularly in higher frequency ranges. This aspect warrants consideration when interpreting parameters sensitive to high-frequency content.

This study provides valuable insights into the agreement between microphone and piezoelectric transducer measurements. While both methods demonstrate utility in capturing acoustic signals, the observed discrepancies highlight the need for careful consideration of the specific parameters of interest and the potential influence of methodological factors. Future research should build upon these findings to further explore the potential and limitations of contact-based methods in acoustic voice analysis, particularly in ecologically valid settings.

CONCLUSION

This study examined the agreement between a microphone and a piezoelectric transducer in the analysis of synthesized vowels. Using Bland–Altman plots, the study compared acoustic parameter measurements, including f_o, jitter, shimmer, noise, and spectral characteristics. The findings indicated that the degree of agreement between the two methods varied depending on the specific acoustic parameter. While the results support the study's central hypothesis of moderate to high agreement between the acoustic parameters of the two devices, discrepancies were observed. Specifically, the microphone tended to record slightly higher values for most parameters, whereas the transducer showed greater sensitivity for specific parameters such as tilt and shimmer.

The observed differences in sensitivity and the moderate variability for certain parameters underscore the need to take the specific characteristics of each measurement method into account. While both the microphone and the transducer can be used for acoustic voice analysis, the observed discrepancies indicate that neither method can be universally interchanged with the other. The selection of a measurement method should be determined by the specific requirements of the research or clinical application, with careful consideration given to the precision required for individual parameters. For instance, parameters related to f_o demonstrated excellent agreement, suggesting interchangeability for these measures. Conversely, parameters with moderate agreement, such as shimmer and spectral slope, call for more cautious interpretation when comparing data obtained from the two different methods.

In light of the findings, it is premature to conclude that piezoelectric transducers can be widely used as a direct replacement for microphones in all applications.

This study, conducted using synthesized vowels emitted by a loudspeaker, serves as an initial, preliminary investigation into the agreement between a microphone and a piezoelectric transducer. Its laboratory-based design, while offering control over the acoustic signal, deliberately excluded the complex interactions of sound with human anatomy and physiology. Consequently, while it provides insights into the fundamental response differences between the sensors to a controlled stimulus, these findings are not directly generalizable to human voice analysis.

Future research is essential to determine the agreement between these methods in the analysis of human voice. Such studies should investigate variables like transducer placement on the neck, individual participant characteristics, and different speech tasks, and could benefit from comparing findings from controlled laboratory sources, like studio-grade loudspeakers, with those from human phonation, as part of a phased approach to understanding sensor performance. This presents additional complexities due to the dynamic nature of voice and speech and to potential variations in transducer placement and coupling. Specifically, future studies should investigate the influence of factors such as individual participant characteristics, device positioning, and recording environment on the observed discrepancies. Furthermore, research should focus on developing standardized protocols and correction methods to minimize variability between methods, ultimately enhancing the reliability and comparability of acoustic voice analysis results across different recording techniques. The determination of the clinical utility of piezoelectric transducers as an alternative or complementary tool to microphones in voice assessment will be crucial.

Statement

This work was presented in an oral communication at The Voice Foundation in 2025.

Declaration of Competing Interest

The authors declare that they have no known competing financial interests or personal relationships that could have appeared to influence the work reported in this paper.

Declaration of Generative AI and AI-assisted technologies in the writing process

In accordance with the journal's guidelines, the authors acknowledge the use of generative AI and AI-assisted technologies in the writing process solely for improving readability and language. These tools were applied with human oversight, and all generated content was meticulously reviewed and edited to ensure accuracy and integrity.

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