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Section

Acoustics and Vibrations

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Editor-in-Chief

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Section Information

The “Acoustics and Vibrations” Section of Applied Sciences is open to receive high quality original research and review articles related to all aspects and applications of acoustic and elastic waves, as well as vibrations both in their active and passive form. This includes cases where waves are deliberately excited for monitoring purposes, as well as study of seismic waves or acoustic emissions originating from crack propagation within a material, or acoustic conditions in a concert hall. The focus is on the development of innovative techniques and material properties or processes monitored.

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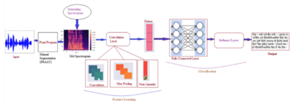
Section Acoustics and Vibrations

Featured Papers

DOI:10.3390/app12126223

Developing a Speech Recognition System for Recognizing Tonal Speech Signals Using a Convolutional Neural Network

Authors: Sakshi Dua, Sethuraman Sambath Kumar, Yasser Albagory, Rajakumar Ramalingam, Ankur Dumka, Rajesh Singh, Mamoon Rashid, Anita Gehlot, Sultan S. Alshamrani and Ahmed Saeed AlGhamdi

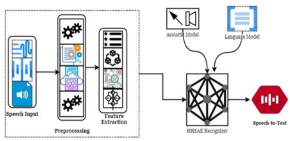


Abstract: Deep learning-based machine learning models have shown significant results in speech recognition and numerous vision-related tasks. The performance of the present speech-to-text model relies upon the hyperparameters used in this research work. In this research work, it is shown that convolutional neural networks (CNNs) can model raw and tonal speech signals. Their performance is on par with existing recognition systems. This study extends the role of the CNN-based approach to robust and uncommon speech signals (tonal) using its own designed database for target research. The main objective of this research work was to develop a speech-to-text recognition system to recognize the tonal speech signals of Gurbani hymns using a CNN. Further, the CNN model, with six layers of 2DConv, 2DMax Pooling, and 256 dense layer units (Google's TensorFlow service) was also used in this work, as well as Praat for speech segmentation. Feature extraction was enforced using the MFCC feature extraction technique, which extracts standard speech features and features of background music as well. Our study reveals that the CNN-based method for identifying tonal speech sentences and adding instrumental knowledge performs better than the existing and conventional approaches. The experimental results demonstrate the significant performance of the present CNN architecture by providing an 89.15% accuracy rate and a 10.56% WER for continuous and extensive vocabulary sentences of speech signals with different tones.

DOI:10.3390/app12031091

Harris Hawks Sparse Auto-Encoder Networks for Automatic Speech Recognition System

Authors: Mohammed Hasan Ali, Mustafa Musa Jaber, Sura Khalil Abd, Amjad Rehman, Mazhar Javed Awan, Daiva Vitkutė-Adžgauskienė, Robertas Damaševičius and Saeed Ali Bahaj

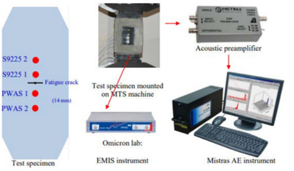


Abstract: Automatic speech recognition (ASR) is an effective technique that can convert human speech into text format or computer actions. ASR systems are widely used in smart appliances, smart homes, and biometric systems. Signal processing and machine learning techniques are incorporated to recognize speech. However, traditional systems have low performance due to a noisy environment. In addition to this, accents and local differences negatively affect the ASR system's performance while analyzing speech signals. A precise speech recognition system was developed to improve the system performance to overcome these issues. This paper uses speech information from jim-schwoebel voice datasets processed by Mel-frequency cepstral coefficients (MFCCs). The MFCC algorithm extracts the valuable features that are used to recognize speech. Here, a sparse auto-encoder (SAE) neural network is used to classify the model, and the hidden Markov model (HMM) is used to decide on the speech recognition. The network performance is optimized by applying the Harris Hawks optimization (HHO) algorithm to fine-tune the network parameter. The fine-tuned network can effectively recognize speech in a noisy environment.

DOI:10.3390/app12031372

An Artificial Intelligence Approach to Fatigue Crack Length Estimation from Acoustic Emission Waves in Thin Metallic Plates

Authors: Joseph Chandler Garrett, Hanfei Mei and Victor Giurgiutiu

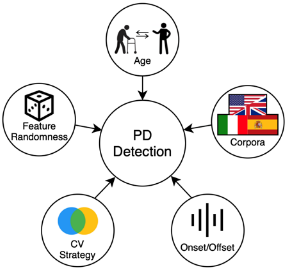


Abstract: The acoustic emission (AE) technique has become a well-established method of monitoring structural health over recent years. The sensing and analysis of elastic AE waves, which have involved piezoelectric wafer active sensors (PWAS) and time domain and frequency domain analysis, has proven to be effective in yielding fatigue crack-related information. However, not much research has been performed regarding (i) the correlation between the fatigue crack length and AE signal signatures and (ii) artificial intelligence (AI) methodologies to automate the AE waveform analysis. In this paper, this crack length correlation is investigated along with the development of a novel AE signal analysis technique via AI. A finite element model (FEM) study was first performed to understand the effects of fatigue crack length on the resulting AE waveforms and a fatigue experiment was performed to capture experimental AE waveforms. Finally, this database of experimental AE waveforms was used with a convolutional neural network to build a system capable of performing automated classification and prediction of the length of a fatigue crack that excited respective AE signals. AE signals captured during a fatigue crack growth experiment were found to match closely with the FEM simulations. This novel AI system proved to be effective at predicting the crack length of an AE signal at an accuracy of 98.4%. This novel AI-enabled AE signal analysis technique will provide a crucial step forward in the development of a comprehensive structural health monitoring (SHM) system.

DOI:10.3390/app12030991

Things to Consider When Automatically Detecting Parkinson's Disease Using the Phonation of Sustained Vowels: Analysis of Methodological Issues

Authors: Alex S. Ozbolt, Laureano Moro-Velazquez, Ioan Lina, Ankur A. Butala and Najim Dehak



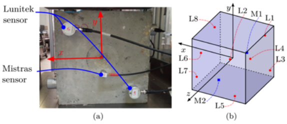
Abstract: Diagnosing Parkinson's Disease (PD) necessitates monitoring symptom progression. Unfortunately, diagnostic confirmation often occurs years after disease onset. A more sensitive and objective approach is paramount to the expedient diagnosis and treatment of persons with PD (PwPDs). Recent studies have shown that we can train accurate models to detect signs of PD from audio recordings of confirmed PwPDs. However, disparities exist between studies and may be caused, in part, by differences in employed corpora or methodologies. Our hypothesis is that unaccounted covariates in methodology, experimental design, and data preparation resulted in overly optimistic results in studies of PD automatic detection employing sustained vowels. These issues include record-wise fold creation rather than subject-wise; an imbalance of age between the PwPD and control classes; using too small of a corpus compared to the sizes of feature vectors; performing cross-validation without including development data; and the absence of cross-corpora testing to confirm results. In this paper, we evaluate the influence of these methodological issues in the automatic detection of PD employing sustained vowels. We perform several experiments isolating each issue to measure its influence employing three different corpora. Moreover, we analyze if the perceived dysphonia of the speakers could be causing differences in results between the corpora. Results suggest that each independent methodological issue analyzed has an effect on classification accuracy. Consequently, we recommend a list of methodological steps to be considered in future experiments to avoid overoptimistic or misleading results.

DOI:10.3390/app12083918

Analysis of Acoustic Emission Activity during Progressive Failure in Heterogeneous Materials: Experimental and Numerical Investigation

Authors: Leandro Ferreira Friedrich, Boris Nahuel Rojo Tanzi, Angélica Bordin Colpo, Mario Sobczyk, Giuseppe Lacidogna, Gianni Niccolini and Ignacio Iturrioz

Abstract: This work focuses on an experimental and numerical investigation into monitoring damage in a cube-shaped concrete specimen under compression. Experimental monitoring uses acoustic emission (AE) signals acquired by two independent measurement apparatuses, and the same damage process is numerically simulated with the lattice discrete element method (LDEM). The results from the experiment and simulation are then compared in terms of their failure load, final configurations, and the evolution of global parameters based on AE signals, such as the b-value coefficient and the natural time approach. It is concluded that the results from the AE analysis present a significant sensitivity to the characteristics of the acquisition systems. However, natural time methods are more robust for determining such differences, indicating the same general tendency for all three data sets.



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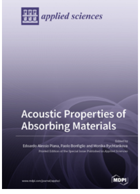

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