Abstract

A method for independently modifying the time and pitch scale of acoustic signals, with an emphasis on speech signals is proposed in this paper. The algorithm developed here is based on Short Time Fourier Transform (STFT). The purpose of this paper is to devise a way to change the rate of a pre-recorded sound without altering the frequency content. Simply playing the sound at a different rate is not a solution. The frequencies would be distorted in proportion
to the scaling factor, and at very low or high rates, would be very difficult to understand at all, let alone identify as human speech. Our approach is to sample the digital signal and then interpolate data points between our samples to produce a sound of the desired length. A slowed-down sound would have more points inserted between the samples than the original signal had, and a speeded-up sound would have fewer than the original. Performance of the proposed algorithm is demonstrated using spectrum plots.

References


Index Terms

Computer Science
Computing, Communication
And Sensor Network
Keywords
Speech Analysis  Speech Processing  Time Scale Modification  Wavelet Packet Transform