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THE SIMULATION WAVELET BASED ANALYSIS FOR NATURAL SOUNDING SIGNAL PROCESS

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ABSTRACT: Speech in one of most important things in our life. Creating community between human and machine using speech is one of critical problems. Signal processing helps to solve the basic stages of various areas of this problem. Working in the field of speech-audio signals processing is conducted very actively. Article describes examples of analysis and speech-audio signals one of the methods processing.

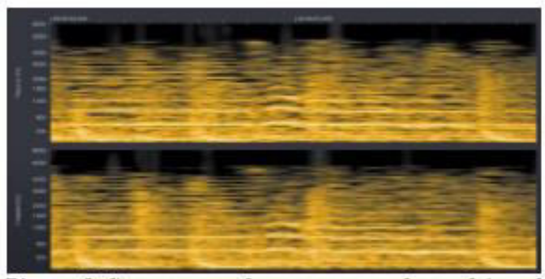
Keywords: acoustic signals, frequency range, identification, signal processing, wavelet and spectrum analysis.

I. INTRODUCTION

The audio is a combination of tones with different pitches and timbres along with noise. In the case of speech signals therefore the most important question is the identification of transmitted information inside the perceived tones combination. So as to understand the entity of this problem, it's required take the physical nature of tones into consideration. All natural sounds likewise speech signals can be described as an infinite sum of sine waves. The frequency of a sine wave is related to what we sense as pitch. Because humans perception of tones is really physiological possible only in the range from 20 Hz up to 20.000 Hz, the mentioned infinite sum turns into a finite sum which is possible to handle only on specially technical equipment, also with according computer programs. Most computer programs are targeted to analyze and interpret the time and frequency content of acoustic signals. Typically analysis types in those computer programs are spectrum analysis, wavelet analysis, spectral analysis and spectral histogram.

II. SPECTRUM ANALYSIS

Spectrum analysis present each acoustic signal as sum of frequency components in a fixed frequency range. Conversion of the time series will be realize with the mathematical tool Fast Fourier Transform (FFT). The spectrogram will be created as spectra of slices of acoustic signal at regular intervals and shows the signal frequencies for the time of signal duration (Picture 1):



Picture 1. Spectrogram for a compressed sound (vocal with instrumental accompaniment)

The graphical mapping of spectrogram shows the frequent ranges, where the most of sound energy is concentrated. Whereat the frequency and time resolution can be modified by the

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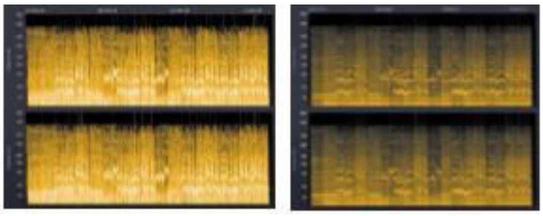
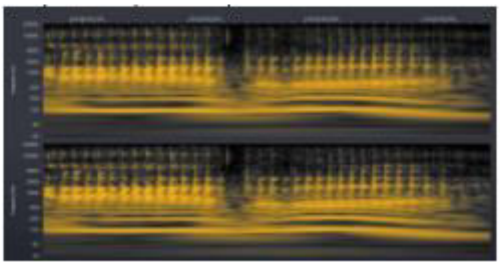
length determination of each slice. Short time slices give a better resolution for the time duration, but will blur the resolution of frequencies. Longer time slices give a better frequency resolution with blurred time resolution (Picture 2).



Picture 2. Spectrogram with short (left) and long (right) slices for the same sound.

III. WAVELET ANALYSIS

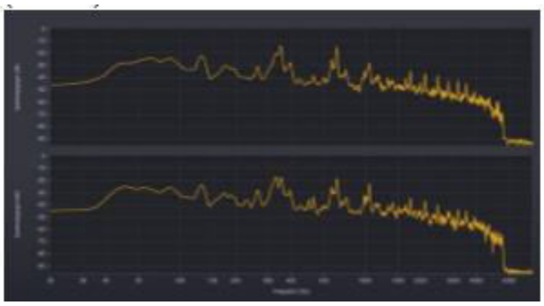
The wavelet analysis is similar to the spectrogram and also present each acoustic signal as sum of frequency components in a fixed frequency range. For Wavelet conversion of the time series certainly is used the wavelet transform. The most different to spectrogram is the dependence of the slice length from the frequency. In result higher frequencies of the signal get a better time resolution, but with blurred resolution of lower frequencies (Picture 3).



Picture 3. Wavelet for a non-compressed speech signal.

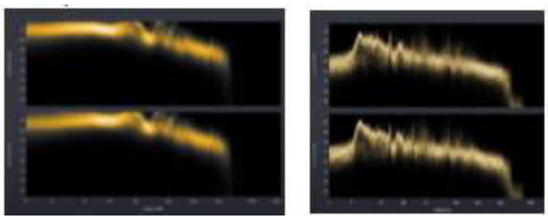
For analysis of acoustic signals the information about the loudness timeline is not representative for any deduction. More important however is the correlation between sound intensity and the frequencies.

So spectral analysis shows, how is depending the intensity from the frequencies inside the signal. In the result of analysis can be determined the distribution range of frequencies (Picture 4).



Picture 4. Spectral analysis for a compressed sound (vocal with instrumental accompaniment).

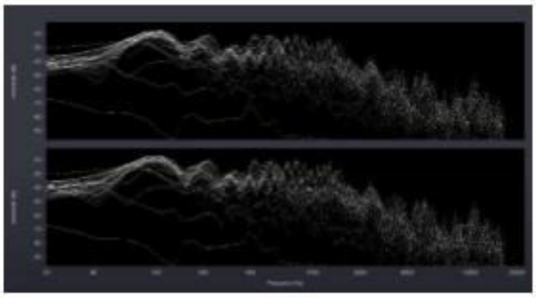
Whereat the frequency and intensity resolution can be modified by the length determination of each slice. Short slices blur the intensity and long slices blur the frequencies (Picture 5).



Picture 5. Spectral analysis with short (left) and long (right) slices for a compressed sound (vocal with instrumental accompaniment).

IV. SPECTRAL HISTOGRAM

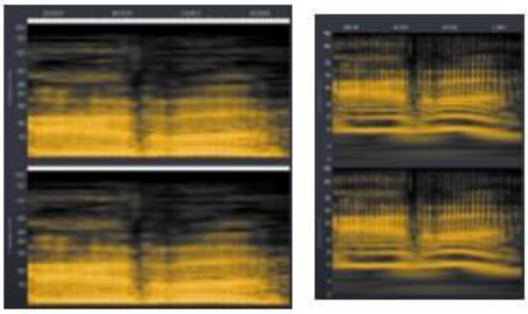
The spectral histogram shows the distribution of the amplitude levels (loudness) per frequency component for the time of signal duration. All Amplitude levels that occurred more frequently will have a brighter intensity (Picture 6). Within the spectral analysis can be designated some correlation levels between intensity and frequencies characterized by lines. The resolution for frequencies and intensity also can be modified by the length determination of slices. The result will be analogues as by the spectral analysis.



Picture 6. Spectral histogram for a non compressed speech signal

Most of compression methods are cropping the frequencies of acoustic signals. In result higher frequencies (over 8.000 Hz) are non available for analysis. This principle is also in use by electronic communication systems. For musical sounds those compression has not negative effects.

The quality of music reproduction in this cases is no essential perceptible. For communication those compression of the signal also mostly is unproblematic. The human brain has the property to compensate missing components of information in acoustic signals and independent complete them by generic context. Certainly this is valid only for mother language and foreign languages at native level. In all other cases independent compensation and completion probable is not possible. Analogues problems exists also by identification of signals in electronic systems. Non compressed signals certain also information for frequencies higher than 8.000 Hz up to 20.000 Hz (Picture 7). By this reasons non compressed signals have some advantages in comparison with compressed signals. It concerns especially the explicit identification of each received signal. How have shown practically researches, especially in speech signals exists phonemes, that have the most part of components in high frequently range, regardless of their perception like a low frequently signal.



Picture 7. Spectrogram (left) and wavelet (right) for a non compressed speech signal.

The absence of signal components higher 8.000 Hz can have the consequence, that the identification process is not unequivocally and in extreme cases cannot be compensated. The consequence is the presence of some alternate interpretations for the signal. Accordingly possible various reactions. Important question in this context is the quantity of required attributes for responsible identification of the information in every acoustic signal. Furthermore in all cases must be considerate the influences of loudness, speed, diction and intonation to the value of each attribute. Higher loudness always will enhance the frequencies in the signal. Faster speed is negative affecting the time resolution of the signal and often slur up to full syllables. Accurate diction always can correct the identification process. Right intonation positive affecting the identification. All executed tests with speech signals have shown, that the divergences in equal signals with various spokespersons minimal influenced the attributes and scarcely have had any importance for the identification process.

V. CONCLUSION

For solution of such tasks still were used methods of mathematical analysis and identification of speech. Because of frequency range reduction during transfer of speech signal individual characteristics of user's speech cannot be completely used.

Proposed solutions are designed to exclude these mistakes and represent absolutely new approach for solution of speech signal identification problems

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