

The Constant-Q Transform Spectral Envelope Coefficients: A Timbre Feature Designed for Music

I. SCOPE

TIMBRE is the attribute of sound which makes, for example, two musical instruments playing the same note sound different. It is generally associated with the spectral (but also temporal) envelope and is typically assumed to be independent from the pitch (but also the loudness) of the sound [1]. In this article, we will show how to design a simple but efficient pitch-independent timbre feature which is well-adapted to musical data, by deriving it from the constant-Q transform (CQT) [3], [4], a log-scaled frequency transform which matches the notes of the Western music scale. We will show how to decompose the CQT into an energy-normalized pitch component and a pitch-invariant spectral envelope, the latter from which we will extract a number of meaningful coefficients. We will then evaluate the discriminative power of these CQT spectral envelope coefficients (CQT-SEC) on the NSynth dataset [5], a large-scale dataset of musical notes which is publicly available, comparing them with the mel-frequency cepstral coefficients (MFCCs) [2], features originally designed for speech recognition but commonly used to characterize timbre in music.

II. RELEVANCE

A timbre feature which is well-adapted to musical data, pitch-independent, and with high discriminative power can be helpful for a number of applications, such as computing similarity, allowing identification, and performing classification of musical data, in particular, musical instruments. Additionally, the ability to decompose the spectrum of a sound (here, the magnitude CQT) into a pitch-independent spectral envelope and an energy-normalized pitch component, can be useful for music analysis, transformation, and resynthesis. The energy-normalized pitch component can also potentially be used for pitch identification and melody extraction.

III. PREREQUISITES

Basic knowledge of audio signal processing and music information retrieval (MIR) is required to understand this article, in particular, concepts such as the Fourier transform, convolution, spectral envelope, pitch, CQT, and MFCCs.

IV. PROBLEM STATEMENT

The complex nature of timbre makes it an attribute that is not easy to quantify in terms of one single characteristic feature. While it is assumed to be independent from pitch and loudness, it is not really possible to fully disentangle timbre from those qualities, as timbre is inherently dependent on the spectral content of the sound, which is also defined by its pitch

and loudness [1]. Researchers in MIR proposed a number of descriptors to characterize one or more aspects of timbre, but ... mostly adopted the MFCCs

V. SOLUTION

convolution theorem: [6].

A. Observations

Assumption: A log-spectrum, such as the CQT-spectrum, can be represented as the convolution of a pitch-invariant log-spectral envelope component (= timbre) and an envelope-normalized pitch component.

- A pitch change in the audio translates to a linear shift in the log-spectrum.
- The Fourier transform (FT) of a convolution of two functions is equal to the point-wise product of their FTs (convolution theorem).
- The magnitude FT is shift-invariant.

VI. NUMERICAL EXAMPLE

VII. WHAT WE HAVE LEARNED

We have shown that ...

VIII. AUTHOR

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