

Review of

Exploring expressive performance trajectories: six famous pianists play six Chopin pieces

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In this paper, the author gave an exploratory method to analyze large amount of expressive performance. They record loudness and tempo as two-dimensional data and was segmented into musically relevant phrases, normalized, and smoothed. This method would give insights of individual expressive strategies and also common performance strategies between individual performers.

There are three parameters defined in the analysis method: the type of normalization applied to the data, the degree of smoothing, and the weighting between tempo and loudness.

Five normalizations could be taken as 3 levels: no normalization is applied at the first level; compare absolute changes with each other at the 2nd level, and compare we compare relative changes.

Five different smoothing levels: none, and smoothing windows corresponding to mean performed durations of $\frac{1}{2}$, 1, or 2, beats either side.

We can change the tempo-loudness weighting as any value that is reasonable.

For the un-normalized data comparison, the segments of a particular piece fall into certain clusters that are different from those of other pieces. So we can tell the one piece from another. Certain expressive strategies that distinguish the pianists are also observed in the visualization shown in Fig.1.

Since we would like to compare phrase segments with different basic tempo and loudness, we got to normalize them. The author showed only "dividing by local mean" normalization and compare the expressive strategies of different pianist. The strategies of those 6 pianists have some part in common, as illustrated in Fig. 6.

This method has the advantage that could deal with large amounts of data, and it reduces complexity and visualizes the data with an interactive interface. The worst problem is, the data have to be verified manually if it has musical relevance. The reliability is still left in question. The author also mentioned four problems of this method. I think we could measure the loudness of each individual sound source instead of measuring the overall loudness. I think there might be no good solution for the data interpolation problem. I've been thinking about something like we should sample all the music files with same amount of data, which means the sampling period is shorter for shorter performance and longer performance. Then I realized it is a basic problem of sampling (A to D) that it might not be solvable right now.

I think this paper provide a possibility that we could find a way to deal large amount of data and find some theory or model. I think this is admirable since all those papers I saw in last a few weeks only deal with few data and I think their conclusion could only be partially right and partially wrong.