

A Computationally Efficient Multipitch Analysis Model

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Aim

- * To design a computationally efficient model for multi pitch and periodicity analysis for audio signals.
- * We demonstrate a two pitch analysis model by T. Tolonen and M. Karjalainen and how this model is more efficient when compared to the unitary pitch analysis model of Meddis and O'Mard.

Applications / importance of the task

- 1) In complex audio signals where complex harmonics of various signals are mixed into a single channel it becomes hard to detect it. This can be done by the use of the multi-pitch analysis model.
- 2) The applications of this analysis include,
 - 1) Sound source separation
 - 2) Structural representation of audio signals
 - 3) Computational auditory scene analysis

Challenges / Motivation

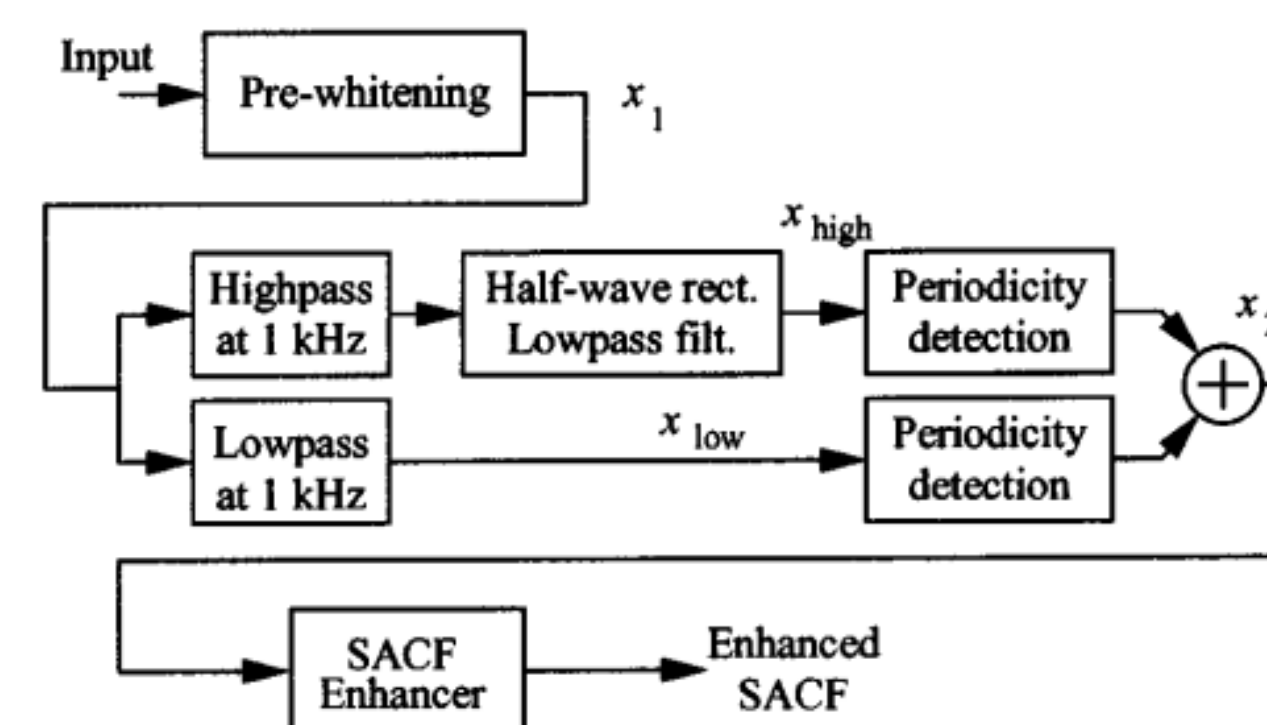
- 1) Although the unitary model has shown good correspondence to human perception, it uses multi-channel filterbank which has 32-120 filters. Calculating the auto-correlation for all the channels makes this model computationally inefficient.
- 2) In this project, we use only 2 channel and also we compute the autocorrelation using DFT. This makes the model computationally more efficient.

References

- [1] Tero Tolonen and Matti Karjalainen. "A computationally efficient multipitch analysis model". In: IEEE transactions on speech and audio processing 8.6 (2000), pp. 708–716.
- [2] Ray Meddis and Lowel O'Mard. "A unitary model of pitch perception". In: The Journal of the Acoustical Society of America 102.3 (1997), pp. 1811–1820.

Method

- * We first use a Hamming window of size 46.4 ms.
- * The second step is similar to the unitary model, the signal is passed through two channels one being a high pass filter and another a low pass filter. This high channel signal is then half-wave rectified and lowpass filtered.
- * The third step involves periodicity detection which is implemented by using the idea of "generalized autocorrelation". It consists of a discrete Fourier transform (DFT), magnitude compression of the spectral representation, and an inverse transform (IDFT).



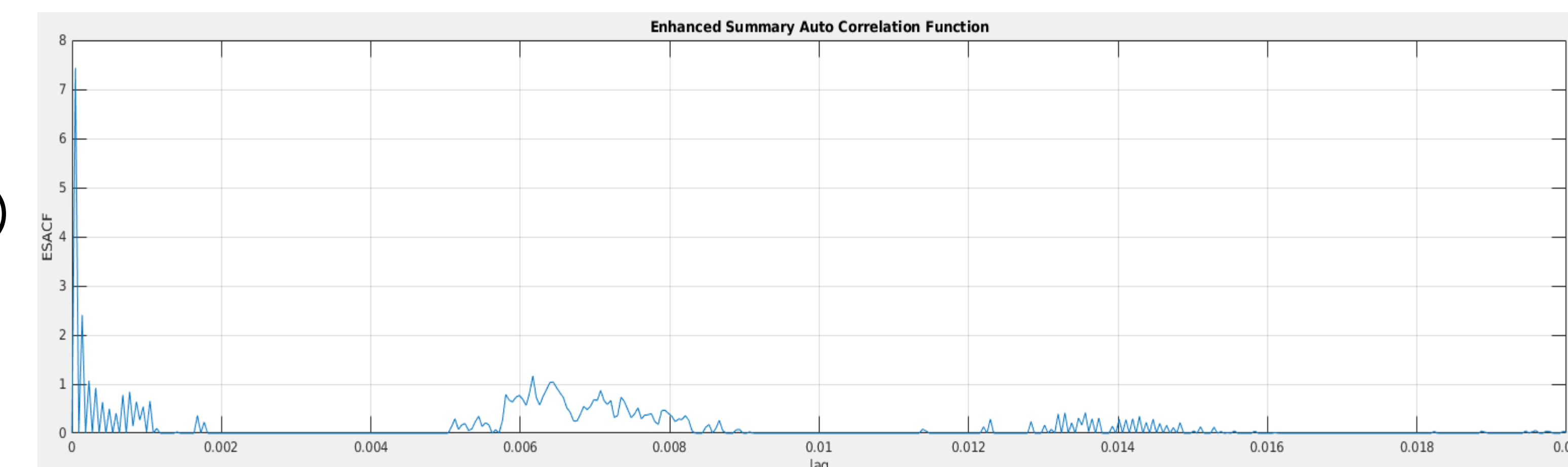
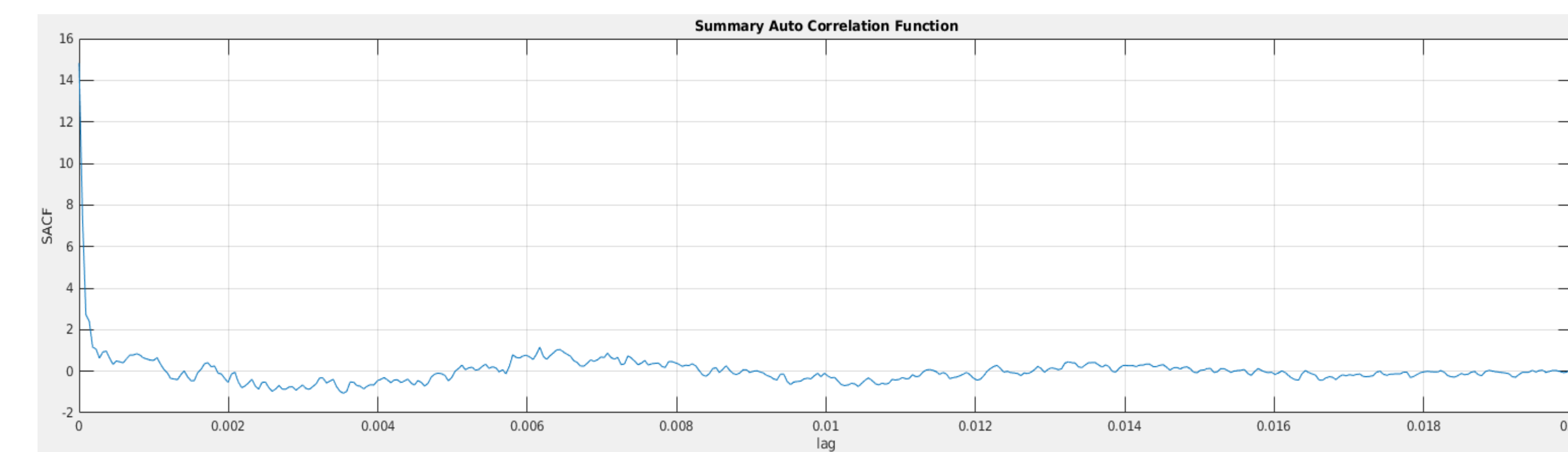
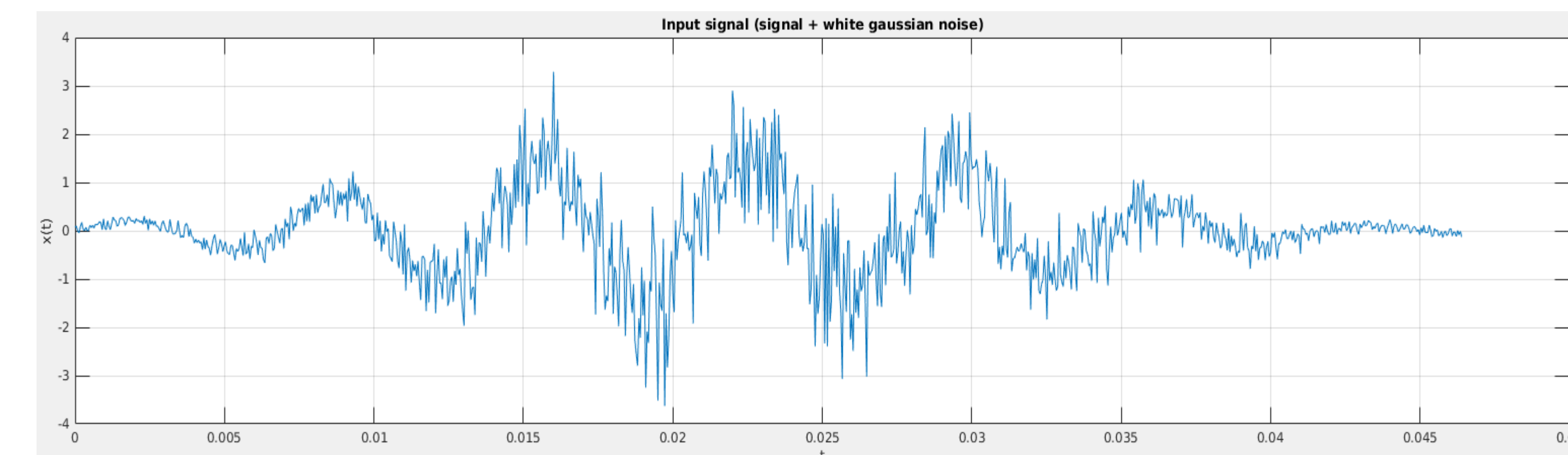
$$x_2 = \text{IDFT}(|\text{DFT}(x_{\text{low}})|^k) + \text{IDFT}(|\text{DFT}(x_{\text{high}})|^k)$$

- * Finally, we enhance the SACF (i.e., x_2) to obtain ESACF.

Parameters used in this model

- * Window size: 46.4 ms
- * Filter order of the high-pass and low-pass: 2
- * $k = 0.67$

MATLAB plots



Conclusion

- * Though the auditory analogy of the model might not be strong, this model shows results from the view of human perception, for example, the pre-whitening filter and the channel filter closely associate to certain phenomena in human perception of sound.
- * This model shows a much more efficient method of multi pitch analysis which negates the main drawback of the unitary model.