Inversion of Auditory Spectrograms, Traditional Spectrograms, and Other Envelope Representations

Envelope representations such as the auditory or traditional spectrogram can be defined by the set of envelopes from the outputs of a filterbank. Common envelope extraction methods discard information regarding the fast fluctuations, or phase, of the signal. Thus, it is difficult to invert, or reconstruct a time-domain signal from, an arbitrary envelope representation. To address this problem, a general optimization approach in the time domain is proposed here, which iteratively minimizes the distance between a target envelope representation and that of a reconstructed time-domain signal. Two implementations of this framework are presented for auditory spectrograms, where the filterbank is based on the behavior of the basilar membrane and envelope extraction is modeled on the response of inner hair cells. One implementation is direct while the other is a two-stage approach that is computationally simpler. While both can accurately invert an auditory spectrogram, the two-stage approach performs better on time-domain metrics. The same framework is applied to traditional spectrograms based on the magnitude of the short-time Fourier transform. Inspired by human perception of loudness, a modification to the framework is proposed, which leads to a more accurate inversion of traditional spectrograms.

Efficient Algorithms for the Discrete Gabor Transform with a Long Fir Window

The Discrete Gabor Transform (DGT) is the most commonly used signal transform for signal analysis and synthesis using a linear frequency scale. The development of the Linear Time-Frequency Analysis Toolbox (LTFAT) has been based on a detailed study of many variants of the relevant algorithms. As a side result of these systematic developments of the subject, two new methods are presented here. Comparisons are made with respect to the computational complexity, and the running time of optimised implementations in the C programming language. The new algorithms have the lowest known computational complexity and running time when a long FIR window is used. The implementations are freely available for download. By summarizing general background information on the state of the art, this article can also be seen as a research survey, sharing with the readers experience in the numerical work in Gabor analysis.
Modulation filtering using an optimization approach to spectrogram reconstruction

Modulations across time and frequency are known from previous studies to play a significant role for speech intelligibility. Hence, well-chosen manipulations of modulations via an accurate tool to systematically modify the modulation content of a signal might be useful for the improvement of speech intelligibility. This study investigates modulation filtering in a time-frequency representation of the signal (e.g., a spectrogram), using a novel approach for reconstructing a signal from its modified representation. It is suggested that this synthesis is regarded as an optimization problem, where the variables are the time samples of the output signal and where the cost function to minimize is the difference between the target spectrogram and the current spectrogram. This approach is made feasible, with regard to the large number of variables involved, by use of a limited-memory optimization algorithm. This study presents basic results regarding temporal modulation filtering and discusses the novel method and its possibilities of improvement.

General information
Publication status: Published
Organisations: Hearing Systems, Department of Electrical Engineering
Contributors: Decorsiere, R. J. B., Søndergaard, P. L., Buchholz, J., Dau, T.
Publication date: 2011

Host publication information
Title of host publication: Proceedings of Forum Acusticum 2011
Publisher: European Acoustics Association
ISBN (Print): 978-84-694-1520-7
Electronic versions:
Decorsiere2011[1].pdf
Source: orbit
Source ID: 282053
Research output: Contribution to journal › Journal article – Annual report year: 2012 › Research › peer-review

On the relationship between multi-channel envelope and temporal fine structure

The envelope of a signal is broadly defined as the slow changes in time of the signal, whereas the temporal fine structure (TFS) are the fast changes in time, i.e. the carrier wave(s) of the signal. The focus of this paper is on envelope and TFS in multi-channel systems. We discuss the difference between a linear and a non-linear model of information-extraction from the envelope, and show that using a non-linear method for information-extraction, it is possible to obtain almost all information about the originating signal. This is shown mathematically and numerically for different kinds of systems providing an increasingly better approximation to the auditory system. A corollary from these results is that it is not possible to generate a test signal containing contradictory information in its multi-channel envelope and TFS.

General information
Publication status: Published
Organisations: Hearing Systems, Department of Electrical Engineering
Contributors: Søndergaard, P. L., Decorsiere, R. J. B., Dau, T.
Publication date: 2011

Host publication information
Title of host publication: Speech Perception and Auditory Disorders
Editors: Dau, T., Christensen-Dalsgaard, J., Jepsen, M. L., Poulsen, T.
ISBN (Print): 978-87-990013-3-0
Electronic versions:
The Linear Time Frequency Analysis Toolbox

The Linear Time Frequency Analysis Toolbox is a Matlab/Octave toolbox for computational time-frequency analysis. It is intended both as an educational and computational tool. The toolbox provides the basic Gabor, Wilson and MDCT transform along with routines for constructing windows (IIR prototypes) and routines for manipulating coefficients. It also provides a bunch of demo scripts devoted either to demonstrating the main functions of the toolbox, or to exemplify their use in speciﬁc signal processing applications. In this paper we describe the used algorithms, their mathematical background as well as some signal processing applications.

General information
Publication status: Published
Organisations: Hearing Systems, Department of Electrical Engineering, Université de Provence, Austrian Academy of Sciences
Contributors: Søndergaard, P. L., Torrésani, B., Balazs, P.
Pages: 1250032
Publication date: 2011
Peer-reviewed: Yes

Towards a binaural modelling toolbox

The Auditory Modelling Toolbox (AMToolbox) is a new Matlab/Octave toolbox for developing and applying auditory perceptual models and in particular binaural models. The philosophy behind the project is that the models should be implemented in a consistent manner, well documented and user-friendly in order to allow students and researchers to actively work with current models and further develop existing ones. In addition to providing the models, it is a goal of the project to collect published human data and denitions of model experiments. This will simplify the verication of models by running the model experiments and comparing the predictions to human data. The software is released under the GNU Public License (GPL) version 3, and can be downloaded from http://amtoolbox.sourceforge.net.

General information
Publication status: Published
Organisations: Hearing Systems, Department of Electrical Engineering, Cardiff University, Technische Universität Berlin, Austrian Academy of Sciences
Publication date: 2011
An Efficient Algorithm for the Discrete Gabor Transform using full length Windows

General information
Publication status: Published
Organisations: Hearing Systems, Department of Electrical Engineering, Centre for Applied Hearing Research
Contributors: Søndergaard, P. L.
Publication date: 2009
Peer-reviewed: Yes

Publication information
Journal: Sampling Theory in Signal and Image Processing
ISSN (Print): 1530-6429
Scopus rating (2009): SJR 0.963 SNIP 1.024
Original language: English
Source: orbit
Source ID: 250417
Research output: Contribution to journal › Conference article – Annual report year: 2009 › Research › peer-review

Finite Discrete Gabor Analysis
Gabor analysis is a method for analyzing signals through the use of a set of basic building blocks. The building blocks consist of a certain function (the window) that is shifted in time and frequency. The Gabor expansion of a signal contains information on the behavior of the signal in certain frequency bands at certain times. Gabor theory can be formulated for both functions on the real line and for discrete signals of finite length. The two theories are largely the same because many aspects come from the same underlying theory of locally compact Abelian groups. The two types of Gabor systems can also be related by sampling and periodization. This thesis extends on this theory by showing new results for window construction. It also provides a discussion of the problems associated to discrete Gabor bases. The sampling and periodization connection is handy because it allows Gabor systems on the real line to be well approximated by finite and discrete Gabor frames. This method of approximation is especially attractive because efficient numerical methods exists for doing computations with finite, discrete Gabor systems. This thesis presents new algorithms for the efficient computation of finite, discrete Gabor coefficients. Reconstruction of a signal from its Gabor coefficients is done by the use of a so-called dual window. This thesis presents a number of iterative algorithms to compute dual and self-dual windows. The Linear Time Frequency Toolbox is a Matlab/Octave/C toolbox for doing basic discrete time/frequency and Gabor analysis. It is intended to be both an educational and a computational tool. The toolbox was developed as part of this Ph.D. project to provide a solid foundation for the field of computational Gabor analysis.

An Efficient Algorithm for the Discrete Gabor Transform using full length Windows
This paper extends the efficient factorization of the Gabor frame operator developed by Strohmer in [1] to the Gabor analysis/synthesis operator. This provides a fast method for computing the discrete Gabor transform (DGT) and several algorithms associated with it. The algorithm is used for the case when the involved window and signal have the same length.

General information
Publication status: Published
Organisations: Department of Mathematics
Contributors: Søndergaard, P. L.
Pages: SPL-04111-2007
Publication date: 2007
Gabor frames by sampling and periodization
By sampling the window of Gabor frame for L-2 (R) belonging to Feichtingers algebra S-0 (R), one obtains a Gabor frame for l(2) (Z). In this article we present a survey of results by R. Orr and A.J.E.M. Janssen and extend their ideas to cover interrelations among Gabor frames for the four spaces L-2 (R), l(2) (Z), L-2 ([O,L]) and C-L. Some new results about the general dual windows with respect to sampling and periodization are presented as well. This theory is used to show a new result of the Kaiblinger type to construct an approximation to the canonical dual window of a Gabor frame for L-2 (R).

Iterative algorithms to approximate canonical Gabor windows: Computational aspects
In this article we investigate the computational aspects of some recently proposed iterative methods for approximating the canonical tight and canonical dual window of a Gabor frame (g, a, b). The iterations start with the window g while the iteration steps comprise the window g, the k(th) itand gamma(k), the frame operators S and S-k corresponding to (g, a, b) and (gamma(k), a, b), respectively, and a number of scalars. The structure of the iteration step of the method is determined by the envisaged convergence order m of the method. We consider two strategies for scaling the terms in the iteration step: Norm scaling, where in each step the windows are normalized, and initial scaling where we only scale in the very beginning. Norm scaling leads to fast, but conditionally convergent methods, while initial scaling leads to unconditionally convergent methods, but with possibly, suboptimal convergence constants. The iterations, initially formulated for time-continuous Gabor systems, are considered and tested in a discrete setting in which one passes to the appropriately sampled-and-periodized windows and frame operators. Furthermore, they are compared with respect to accuracy and efficiency, with other methods to approximate canonical windows associated with Gabor frames.
Iterative algorithms to approximate canonical Gabor windows: Computational aspects

In this paper we investigate the computational aspects of some recently proposed iterative methods for approximating the canonical tight and canonical dual window of a Gabor frame \((g,a,b)\). The iterations start with the window \(g\) while the iteration steps comprise the window \(g\), the \(k\)th iterand \(\gamma_k\), the frame operators \(S\) and \(S_k\) corresponding to \((g,a,b)\) and \((\gamma_k,a,b)\), respectively, and a number of scalars. The structure of the iteration step of the method is determined by the envisaged convergence order \(m\) of the method. We consider two strategies for scaling the terms in the iteration step: norm scaling, where in each step the windows are normalized, and initial scaling where we only scale in the very beginning. Norm scaling leads to fast, but conditionally convergent methods, while initial scaling leads to unconditionally convergent methods, but with possibly suboptimal convergence constants. The iterations, initially formulated for time-continuous Gabor systems, are considered and tested in a discrete setting in which one passes to the appropriately sampled-and-periodized windows and frame operators. Furthermore, they are compared with respect to accuracy and efficiency with other methods to approximate canonical windows associated with Gabor frames.

Symmetric, discrete fractional splines and Gabor systems

In this paper we consider fractional splines as windows for Gabor frames. We introduce two new types of symmetric, fractional splines in addition to one found by Unser and Blu. For the finite, discrete case we present two families of splines: One is created by sampling and periodizong the continuous splines, and one is a truly finite, discrete construction. We discuss the properties of these splines and their usefulness as windows for Gabor frames and Wilson bases.