

Bibliography: Audio Spectral Modeling

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Papers and Books Related to Music 421

This list of references can be used as a starting-point for further reading.

Brief citations appear first, organized by topic, followed by the full citations in alphabetical order. The citations within each topic are generally listed in chronological order, except that the first reference cited may be my recommended best *single* reference (e.g., most recent and/or comprehensive).

Brief Citations Organized by Topic

Elementary Signal Processing: [74, 61, 117, 116, 118, 106, 102]

Intermediate to Advanced Signal Processing: [84, 82, 81, 100, 83, 42, 127]

Fourier Transform Mathematics: [111, 9, 12, 14]

FFT Software: [30, 114]

FFT Windows: [39, 80, 43, 21]

FIR Digital Filter Design: [8, 100, 87, 75, 46, 60]

FIR Hilbert Transformer Design: [46, 103]

Short-Time Fourier Transform (STFT): [4, 2, 16, 40, 3, 95, 78]
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Phase Vocoder: [22, 76, 108, 36, 26, 28, 38, 90, 91, 92, 45, 106, 94]
[50, 52, 51]

Time Compression/Expansion, Pitch Shifting: [94, 96, 50, 53, 52, 51, 119]

Sinusoidal Modeling: [99, 89]

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53, 57, 34, 35, 58, 55, 54, 88, 49, 13]

Sinusoidal Modeling Software: The Spectral Modeling Synthesis (SMS) Home Page¹

Transient Modeling: [1, 56, 53, 57, 58, 55]

Time-Domain Envelope Estimation via LPC: [47, 123]

Spectral Envelope Estimation: [62, 81, 107]

Pitch Estimation: [41, 24, 23]

¹<http://www.iua.upf.es/sms/>

Stochastic spectrum analysis: [44, 126]
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Time and Frequency Scale Modification: [48, 63, 106, 119]
Noise Reduction: [33, 5, 59]
Speech Signal Processing and Coding: [101, 27]
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Texts on Filterbanks: [122, 124]
Multirate background: [122, pp. 100-133] [17] (upsampling,downsampling,multirate identities,polyphase representation, and uniform DFT filter banks)
STFT and filterbank interpretation: [122, Ch. 11,1st two sections,pp. 457–480]
Time-Frequency Distributions, Wigner Distribution: [15]
Aliasing Cancellation: [124, pp. 156-171] [122] [65, pp. 175-193]
Time-Domain Aliasing Cancellation: [93, (original paper)]
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Lapped Orthogonal Transforms (Filterbanks, Audio Coding):
[65, 66, 64]
Perceptual Audio Coding: [10, 120, 19, 93, 66, 79, 6, 11, 58, 7, 86]
Introduction to Wavelets: [104, 124, 121]

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