MUS320A&B: Introduction to Digital Audio Signal Processing

Center for Computer Research in Music and Acoustics (CCRMA)
Department of Music, Stanford University

320A (spectra): Autumn Quarter
320B (filters): Winter Quarter

2015–2016

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Music 320 A & B: Introduction to Digital Audio Signal Processing

1 Course Description

Music 320 is a two-quarter first-course in digital signal processing with applications in computer music and audio.

The lectures present fundamental elements of digital audio signal processing, such as sinusoids, spectra, the Discrete Fourier Transform (DFT), digital filters, z transforms, transfer-function analysis, and basic Fourier analysis in the discrete-time case. Matlab is used for in-class demonstrations and homework/lab assignments. The labs focus on practical applications of the theory, with emphasis on working with waveforms and spectra, "getting sound", and developing proficiency in the matlab language.

Prerequisites: High-school level algebra and trigonometry, some calculus, and prior exposure to complex numbers.

Time and Place

Term:	Autumn and Winter Quarters
Location:	CCRMA Classroom (Knoll 217)
Lectures:	Tuesdays and Thursdays 3:00–4:50 PM
Units:	3–4
Instructor:	Julius O. Smith (jos@ccrma.stanford.edu)
TA:	Iran Roman (iran@ccrma.stanford.edu)
Office Hours:	See "Office Hours and Getting Help"[1] below
Schedule:	See "Schedule and Pointers"[2] below

2 Administrative Information

2.1 Announcements

Class announcements are often made via *email*. For this we are presently using Piazza:

`https://piazza.com/stanford/winter2016/music320b/home`

You should have received an invitation from Piazza to join the class after you signed up for it in axess (using the email address known to axess). Otherwise, please join by visiting the above URL and entering your preferred email address.

[1] `http://ccrma.stanford.edu/~jos/intro320/Office_Hours_Getting_Help.html`
[2] `http://ccrma.stanford.edu/~jos/intro320/`

2.2 Assignments

There are five homework/lab assignments, each covering roughly two weeks of the course. In each two-week "section", the first week is devoted primarily to theory while the second week is focused more on software and applications. Thus, each assignment contains both a theory and laboratory part. The lab portion typically requires programming in matlab.

Each assignment is typically announced on Tuesday in the first week of the section. The theory part is normally due the following Tuesday at 3:15 pm in the 320 mailbox (located in the Knoll, central wing, second floor, facing the printer). The lab part is normally due by midnight the following Friday, i.e., at the end of the two-week section.

For lab assignments, we will be using the Coursework[3] website. To sign up, go to the Coursework website and find Music320B. Once you are enrolled in the class, you can upload your matlab files in the "drop box" on the left menu.

See §2.5 below regarding obtaining help with theory and lab assignments.

Regarding late homeworks, 7 free late days are allowed (with hours rounded up to the nearest day). Late homeworks beyond this will not be accepted. Only up to 3 late days can be used for any one assignment. When using late days, students are required to write the number of late days used at the top of the assignment (date and time).

Students are encouraged to discuss the homework assignments with each other. It is fine to learn from a classmate how to solve any of the homework problems, but each student is responsible for carrying out and writing up the assignments individually. It is an honor code violation to *copy* the work of others.

2.3 Exams

The final examination will be held in the CCRMA Classroom (Knoll 217) on the University-assigned date, also listed for convenience in the class schedule (§**??** on page **??**).

2.4 Grading

Grades are based on the homeworks/labs (60%), and the final exam (40%). There are also bonus points available based on general participation. The weightings may be changed as we see fit.

2.5 Office Hours and Getting Help

We will be using Piazza[4] for sharing answers to theory and lab questions with the whole class. To sign up, see the 320 Piazza site.[5] It is free and allows you to view past questions from other students, and discuss questions together. Try it first for any homework questions you may have. You are also welcome, of course, to catch us whenever you see us at CCRMA, such as during office hours, etc.

TA weekly office hours will be announced in class and via email to the class. Meetings with JOS are arranged via email for half-hour slots before or after class, or other times when necessary.

[3] `https://coursework.stanford.edu`
[4] `https://www.piazza.com`
[5] `https://piazza.com/stanford/winter2016/music320b/home`

3 Textbooks

Music 320A (fall) is based on assigned chapters of

Mathematics of the Discrete Fourier Transform (DFT),[6] by Julius O. Smith

Music 320B (winter) is based on assigned chapters of

Introduction to Digital Filters,[7] by Julius O. Smith

See §**??** for the list of assigned chapters. Both books are fully available on-line. Softcover versions are available from Amazon.com.

4 The Partially Flipped Classroom

With the lectures recorded, class time is freed up for other activities. Here is how a typical "partially flipped class" is organized:

- Q&A session on the reading/video content
- Review of main points in the reading/videos
- Demos in support of the reading/videos
- Presentation of the homework/lab assigment
- Worked problems similar to those in the homework
- Matlab session on theory/lab-related topics
- Live coding in matlab

Additional available time may be devoted to

- More demos
- More discussion
- "Backwards learning" examples:
  - Plugins using spectral techniques
  - Faust language and some of its examples
- More on applications and why all this is useful
- Preview material coming up

[6] `http://ccrma.stanford.edu/~jos/mdft/`
[7] `http://ccrma.stanford.edu/~jos/filters/`

5 A Recipe for Learning

Learning something new requires multiple passes on the material. For example:

1. Do the assigned reading at a fixed pace to get a picture of what's covered
2. Watch the lecture videos, pausing and taking notes on anything newly learned
3. Make a first pass on the homework, flagging and skipping when stuck on a problem
4. Discuss nonobvious homework problems with other students, the TA, and/or JOS
5. Write up the homework problems, everything now understood
6. Exam prep: Reread the text for full comprehension
7. Exam prep: Reread your notes
8. Prepare your one-page summary of the course allowed in the exam
9. Exam experience: Exercise in problem solving using the material

These multiple engagements typically result in a fair amount of learning.

6 320B Schedule and Pointers

6.1 Section 1: Linearity and Time Invariance; Time-Domain Representations

- **Reading:**
  - Chapters 1 and 2 of **Introduction to Digital Filters**[8]
  - Chapter 4 (Linearity and Time Invariance) and Chapter 5 (Time Domain Filter Representations) of **Introduction to Digital Filters**
  - First section of Chapter 9 (Implementation Structures) on the Four Direct Forms
  - Optionally peruse the Music 421 overheads pertaining to acyclic convolution
  - Assignment 1
- **Lecture Videos (Total Viewing Time ≈ 2 Hours):**
  - Linear Time-Invariant (LTI) Filters, Convolution, Ideal Lowpass, Guard Band, Transition Band, Simplest Lowpass Filter, Impulse Response, DTFT, Frequency Response, Amplitude Response, Phase Response, Linear Phase, Sinewave Analysis[9] [38:36]
  - Derivation of Convolution from Linearity and Time-Invariance (LTI) (Superposition) [2015][10] [29:08]
  - Recursive Filters, Simplest Lowpass, Phase Delay, Group Delay[11] [28:01]
  - Supplementary: Faust in the Classroom[12] [41:00]
  - Supplementary: Faust Intro[13] [26:00]
  - Supplementary: Faust Implementation of the Simplest Lowpass Filter[14] [18:22]
  - Simplest RECURSIVE LPF, Pole Gain, PFE, Time-Constant of a Pole, Stability Pole, Bandwidth, Laplace Transform, s-plane poles and zeros, s-plane pole corresponds to exponential[15] [38:37]
  - Direct Form Digital Filters, Transposing a Flow Graph, Transposed Direct Forms 1 and 2, Direct Form 1 Biquad, Direct Form 2 Biquad, Transposed Direct Form 2 Biquad, Interpolated Delay-Line Read, Interpolated Delay-Line Write = Transpose of Read[16] [14:35]
  - Simplest Mechanical LPF: Ideal Mass on Frictionless Surface, Newton's law of motion f=ma, Analog Transfer Function for Driving-Force Input, Velocity Output, Admittance (Mobility) of a Mass[17] [5:31]

[8] https://ccrma.stanford.edu/~jos/filters/filters.html
[9] https://www.youtube.com/watch?v=p19QzBxnhvg
[10] https://www.youtube.com/watch?v=KWhqV95fKRw
[11] https://www.youtube.com/watch?v=r0fg8eZAKGs
[12] https://www.youtube.com/watch?v=2lEt7dsziO0
[13] https://www.youtube.com/watch?v=qE1_UzQZnnM
[14] https://www.youtube.com/watch?v=jNcKGlMHE9A
[15] https://www.youtube.com/watch?v=iJ7mnqhVBfk
[16] https://www.youtube.com/watch?v=qZUcTsHkHBQ
[17] https://www.youtube.com/watch?v=BULkMAst6_U

– Simplest Mechanical LPF: Ideal Mass on Frictionless Surface, Differentiation Theorem for Laplace Transforms, Transfer Function of the Force-Driven Mass: Frequency Response, Poles and Zeros,, Amplitude Response, 6dB per octave roll off, Bode Plot, Harald Bode, Phase Response[18] [27:29]

6.2 Section 2: Analysis of Digital Filters

- **Reading:**
  - Chapter 6 (Z-transform),
  - Chapter 6 (Transfer Function Analysis)
  - Chapter 7 (Frequency Response Analysis)
  - First three sections of Chapter 8 (Pole-Zero Analysis)
  - Remainder of Chapter 9 (Implementation Structures)
  - Appendix D (Laplace Transform Analysis)
  - Appendix E (Analog Filters)
  - Appendix I.3 (Bilinear Transform)
  - Appendix B (Elementary Audio Digital Filters)
- Assignment 2
- **Lecture Videos (Total Viewing Time ≈ 2 Hours):**
  - Z Transform, Poles and Zeros, Graphical Amplitude and Phase Response, Phase Delay, Group Delay[19] [27:49]
  - Simplest Electrical LPF: RC lowpass; RLC Circuits: Resistor Equation V = IR, Capacitor Equation Q = CV, Inductor Equation V = L dI/dt; Kirchhoff Node and Loop Analysis: Kirchhoff Loop Constraint (Sum of voltages around a loop is zero), Kirchhoff Node Constraint (Sum of currents into a node is zero); Voltage Transfer Circuits, Laplace Transform Circuit Analysis, Transfer Function of RC LPF: Pole-Zero Analysis, Impulse Response, Time Constant of Decay, Bode Plot[20] [21:39]
  - Simplest Electrical LPF: RC lowpass, continued; Bode Plot; 3dB Bandwidth[21] [7:45]
  - Bandwidth of a Pole[22] [22:16]
  - Analog Low-Shelf Filters, High Shelf, Peaking Equalizer, Mapping s to z, Bilinear Transform (BLT), BLT Doesn't Alias, BLT Frequency Warping[23] [12:30]
  - Bilinear Transform = special case of Moebius Transformation [DON'T MISS THIS ONE!][24] [2:34]

[18] https://www.youtube.com/watch?v=YWJPqHhjf8c
[19] https://www.youtube.com/watch?v=hze4CwetVNA
[20] https://www.youtube.com/watch?v=dEmmtsN-ka4
[21] https://www.youtube.com/watch?v=MOBH66RyXZw
[22] https://www.youtube.com/watch?v=m4zCmvvKFso
[23] https://www.youtube.com/watch?v=NqYGdcDW3dY
[24] https://www.youtube.com/watch?v=JX3VmDgiFnY

– Bilinear Transform Frequency Scaling, Resonance Preservation; Digitizing an Integrator (Mass), RC Filter, Low Shelf; BLT Stability Preservation[25] [8:51]
– Shelf Filters in Faust[26] [22:25]

6.3 Section 3: Digital Filter Design

- **Reading:**
  – Chapter 10 (Elementary Audio Digital Filters)
  – Chapter 11 (Filters Preserving Phase)
  – Appendix I (Recursive Digital Filter Design)
  – Appendix I.2 (Butterworth Filters)
  – Appendix K (Digital Filtering in Faust and Pd)
  Optional: Audio Signal Processing in Faust
- Assignment 2
- **Lecture Videos (Total Viewing Time ≈ 2 Hours):**
  – Analog Filters Reviewed: Transfer Function, Frequency Response, Power Response; Analog Lowpass Design, Maximally Flat Passband, Butterworth Filters[27] [30:46]
  – Butterworth Filter Properties: Maximum Flatness at Infinity, Low Ringing, Mild Phase Response, Poles on a Circle; Spectral Factorization, Series Biquad Realization, Elliptic Function Filters, Chebyshev Optimality, Remez Exchange (Parks-McLellan), firpm in matlab, cvx for Convex Optimization[28] [35:19]
  – Introduction to Functional Audio Stream (Faust): Simplest Lowpass, Utilities in Faust's filter.lib[29] [37:30]
  – More Faust: Testing filters using faust2octave[30] [27:32]
  – Butterworth Power Response, Analytic Continuation, Butterworth Poles, Matlab `butter()` Function[31] [7:02]
  – Example Butterworth Filter of Order 2, Digitization via Bilinear Transform, Frequency Prewarping[32] [18:49]
  – Digital Filter Design and Implementation in Matlab: Noise Removal via Lowpass Filtering, Create Sinusoid and Noise, Matlab's `butter()`, `filter()`, `fdatool` (Filter Design and Analysis tool), Simulating a Telephone Channel Bandwidth[33] [10:30]

[25] https://www.youtube.com/watch?v=aaGdgf65PsY
[26] https://www.youtube.com/watch?v=9RDC4ylap7E
[27] https://www.youtube.com/watch?v=doDMmZfEfbg
[28] https://www.youtube.com/watch?v=pUtUrzVHF3Q
[29] https://www.youtube.com/watch?v=FEf7dpApd6I
[30] https://www.youtube.com/watch?v=Ao1ZriZi8nY
[31] https://www.youtube.com/watch?v=nhhuAxBUleU
[32] https://www.youtube.com/watch?v=UcTHnf4B5tU
[33] https://www.youtube.com/watch?v=I0g5r0_BgRM

– Matlab: `freqz()`, `freqs()`; Continuous Butterworth Filter Analysis; Converting to Second-Order Sections in Matlab using `tf2sos()`; Viewing Butterworth Poles in Matlab using `zplane()`; Excess Delay at Filter Cutoff; `grpdelay()`; Elliptic Filters using `ellip()`; Ripple; `impz()`; Zero Phase versus Minimum Phase (Pre-Ring versus Post-Ring); Minimum-Delay Property of Minimum-Phase Filters; Partial Fraction Expansion in Matlab using `residue()`, `residuez()`, or `residued()`[34] [31:10]

6.4 Section 4: Quality Factor Q, Allpass Filters, State Space, State Variable Filter

- **Reading:**
  - Appendix E.7 (Quality Factor (Q))
  - Appendix C (Allpass Filters)
  - Laplace Analysis of a Force-Driven Mass
  - State-Space Formulation of Digital Filters
  - State-Variable Filter
  - **Supplementary:** State-Space Introduction in Music 420
  - **Supplementary:** State-Space Canonical Forms
  - **Supplementary:**
    Linkwitz-Riley Crossovers: A Primer
    LinkwitzRiley filter
  - Assignment 8
- **Lecture Videos (Total Viewing Time ≈ 3 Hours):**
  - Quality Factor (Q) of a Resonator[35] [7:12]
  - Complex One-Pole Resonator and its Q; Canonical Form of a Biquad (s-plane); Mechanical and Electrical Resonators; Limiters, Compressors, Expanders[36] [39:30]
  - Filter Decay Time is about Q Periods[37] [3:16]
  - Bilinear Transform Frequency Mapping, Analog Computers, State Space Formulation, Physical Derivation of Bilinear Transform, State Variable[38] [38:29]
  - Minimum Phase Filters and Signals; Allpass Filters: Poles and Zeros, Graphical Amplitude and Phase Response, Biquad Realization, Phasing; Allpass-Minimum-Phase Decomposition[39] [28:44]
  - Allpass Filters in z and s Planes; Instability as Noncausality; Laurent Series; Bilateral DTFT; Cepstrum; Converting Arbitrary Spectra to Minimum-Phase Form[40] [39:13]

[34] https://www.youtube.com/watch?v=MxxDS01Ea5o
[35] https://www.youtube.com/watch?v=V04yxnqBuYk
[36] https://www.youtube.com/watch?v=zeClzrKUfQU
[37] https://www.youtube.com/watch?v=kPKZQ16EdcU
[38] https://www.youtube.com/watch?v=GRpAqeVbUWs
[39] https://www.youtube.com/watch?v=Cj6Vjp6k7NM
[40] https://www.youtube.com/watch?v=nCwj1VeXQ44

- Repeated Poles at $s = 0$
  * One Pole at DC in the s Plane[41] [14:17]
  * Mechanical Integrator using a Mass[42] [3:46]
  * Integrator made by a Spring or Inductor[43] [5:46]
  * One Pole at DC in the s Plane, Continued[44] [1:31]
  * Frequency Response of an Integrator[45] [5:05]
  * Repeated Poles at DC[46] [4:16]
  * General Transfer Function of a Pile of Poles at DC[47] [3:38]
  * Impulse Response of a Pile of Poles at DC[48] [3:22]
- The *State Space* Formulation of Linear Systems [2016 flipped-class review]
  * Adding Feedback around the Integrator Chain, Derivation of the State Space Formulation[49] [10:46]
  * State Space Formulation, Continued[50] [1:33]
  * State Space Overview[51] [11:50]
  * Force Driven Mass[52] [4:49]
  * General Discussion of State Space[53] [8:48]
  * Defining State Variables[54] [4:51]
  * State Variable Choice Summary[55] [5:14]
  * General State Space Model and Digitization via Backward Euler[56] [5:16]
  * State Space Converts Nth-Order to Vector First-Order[57] [2:08]
- Moog VCF[58] [23:29]
- Moog VCF Live-Coded in Faust
  * Moog VCF Live-Coded in Faust[59] [30:49]
  * Moog VCF in Faust, Review[60] [5:02]

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[41] `https://www.youtube.com/watch?v=DIhH2JIWQeE`
[42] `https://www.youtube.com/watch?v=c1TIX2Ybn3U`
[43] `https://www.youtube.com/watch?v=qPLyMge71F4`
[44] `https://www.youtube.com/watch?v=lqXYMuna3yw`
[45] `https://www.youtube.com/watch?v=VLtpw4eD4Uc`
[46] `https://www.youtube.com/watch?v=5s6umF7I4T4`
[47] `https://www.youtube.com/watch?v=6_VRX5Fvdig`
[48] `https://www.youtube.com/watch?v=d6uQYhQteaw`
[49] `https://www.youtube.com/watch?v=a__oM8rYPHc`
[50] `https://www.youtube.com/watch?v=ZNO2687Nch0`
[51] `https://www.youtube.com/watch?v=Ygq66m9NrDk`
[52] `https://www.youtube.com/watch?v=zUibjGb6QQ8`
[53] `https://www.youtube.com/watch?v=F0ku7LwHrI0`
[54] `https://www.youtube.com/watch?v=45QwWhbJMvU`
[55] `https://www.youtube.com/watch?v=LgggKFhh0uo`
[56] `https://www.youtube.com/watch?v=rbrJnknh_dU`
[57] `https://www.youtube.com/watch?v=zdO2nQKeaAY`
[58] `https://www.youtube.com/watch?v=KxBBcNWbZHY`
[59] `https://www.youtube.com/watch?v=WLvpGN_UNlA`
[60] `https://www.youtube.com/watch?v=VV2eOChjTrc`

    * Moog VCF in Faust, Frequency Responses[61] [7:39]
    * [Can Skip] Q-Correction and Gain-Correction Tables[62] [3:38]
  - State Variable Lowpass, Bandpass, and Highpass
    * Normalized Biquad Lowpass Filter, Continuous Time[63] [19:29]
    * State Variable Realization of Normalized Biquad Lowpass[64] [31:22]
    * State Variable Filter Lowpass/Bandpass/Highpass[65] [3:00]
    * State Variable Filter LP/BP/HP Frequency Scaling and Digitization[66] [11:35]
  - Note on Repeated Poles[67] [4:40]

6.5 Section 5: Voice Synthesis, F0 Estimation, Cepstra, Converting to Minimum Phase

- **Reading:**
  - Chapter 12 (Minimum Phase Digital Filters)
  - **Supplementary:** Spectral Envelopes via Cepstrum or LPC [from Music 421 overheads]
- **Lecture Videos:**
  - Complex and Real Cepstrum, Quefrency[68] [7:54]
  - Mel Frequency Cepstral Coefficients (MFCC); Bark and Equivalent Rectangular Bandwidth (ERB) Psychoacoustic Frequency Scales based on Critical Bands of hearing[69] [7:54]
  - Complex Cepstrum Derived; Converting Mixed-Phase Signals to Minimum Phase[70] [5:39]
  - Series Expansion of Log of 1/(1-x)[71] [7:57]
  - Series Expansion of Log Transfer Function in Factored Form[72] [5:26]
  - Contribution of Zeros to the Complex Cepstrum[73] [3:15]
  - Contribution of Poles and Zeros (Inside and Outside the Unit Circle) to the Complex Cepstrum; Nonparametric Cepstral Folding Method for Converting Mixed Phase to Minumum Phase using the FFT; Testing for Time Aliasing[74] [9:14]

[61] https://www.youtube.com/watch?v=cGuuG1bp1JI
[62] https://www.youtube.com/watch?v=QOm0njDsQYY
[63] https://www.youtube.com/watch?v=hjHPMbDa3yk
[64] https://www.youtube.com/watch?v=9cXaEIOeWuU
[65] https://www.youtube.com/watch?v=o-Zpot5TKoE
[66] https://www.youtube.com/watch?v=LGzSnwTFB-0
[67] https://www.youtube.com/watch?v=D6_AK7mfQnQ
[68] https://www.youtube.com/watch?v=Tlk6CLclPPU
[69] https://www.youtube.com/watch?v=2O1BBFeQGOE
[70] https://www.youtube.com/watch?v=0H1z4OW-7UA
[71] https://www.youtube.com/watch?v=wrslb9U7HaI
[72] https://www.youtube.com/watch?v=x3gCA_rj13k
[73] https://www.youtube.com/watch?v=niY8EA4-peA
[74] https://www.youtube.com/watch?v=-_yAOlUTCko

- Nonparametric Cepstral Folding Method in Matlab: `minphasespec()`, `fold()`, `invfreqz()`[75] [6:11]
- Review of another Cepstral Folding Example[76] [19:08]
- Minimum Phase Conversion by Spectral Factorization or Cepstral Method[77] [5:45]
- Minimum Phase Conversion by the Cepstral Method, Continued[78] [12:10]
- Cepstral Method Code, then CollideFX Demo by Chet Gnegy[79] [55:51]
- Voice Vowel Synthesis in Faust[80] [11:26]
- Voice Vowel Synthesis in Faust, Continued[81] []

6.6 Final Exam

Final Exam: Tuesday, March 15, 2016, 7:00-10:00 PM, in the CCRMA Classroom

- The exam will cover readings, homework problems, and laboratory assignments.
- The exam will be *closed book*, except that you may bring an 8.5” by 11” sheet of paper, covered front and back with handwritten notes.
- No calculators allowed (you shouldn’t need one).

[75] `https://www.youtube.com/watch?v=PTx0capBmHU`
[76] `https://www.youtube.com/watch?v=V7K4rmT94PE`
[77] `https://www.youtube.com/watch?v=7GcCkMcqVao`
[78] `https://www.youtube.com/watch?v=5RNeaaFxfVg`
[79] `https://www.youtube.com/watch?v=nkt3bRO4tI4`
[80] `https://www.youtube.com/watch?v=GR97SMvS4Fw`
[81] `https://www.youtube.com/watch?v=`