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
2017–2018

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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.

2 Time and Place

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

2 Administrative Information

2.1 Announcements

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


If you signed up for the class in axess before the first day of classes, you should receive an invitation from Piazza to join the class (using the email address known to axess). Otherwise, please join by visiting the above URL and entering your preferred email address.
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 Thursday during the first week of the section. The theory part is normally due the following Thursday at class time, 3 pm, in the 320 mailbox at CCRMA. The lab part is normally due by midnight the following day, i.e., at the end of the two-week section. Available homework time after turning in the assignments should be devoted to viewing the lecture videos for the next section.

For lab assignments, we will be using the Canvas website. To sign up, go to the Canvas website and find Music320. 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 be penalized at 5% per day. When using late days, 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 (§6 on page 4).

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 for sharing answers to posted questions with the whole class. To sign up, see the 320 Piazza site. 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 are Wednesday evenings 7:30-9:30 PM in the CCRMA Ballroom (2nd floor). Meetings with JOS are arranged via email for half-hour slots after class, or other times when necessary.

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3 https://canvas.stanford.edu
4 https://www.piazza.com
5 https://piazza.com/stanford/winter2018/music320b/home
2.6 Computer Usage

Lab exercises will be computer based. All students may obtain a computer account at CCRMA in order to use the computer facilities. It is also possible to work entirely on your own computer, as long as you have the necessary software. However, note that some course materials are restricted to on-campus access, so you should have at least one Stanford computer account from which you access those.

Here is how to obtain a CCRMA computer account:

https://cm-knoll.stanford.edu/usersignup

*Note:* This link only works at CCRMA.

Once you have your account, please log in at CCRMA and take a look at the User’s guide\(^6\) tab in the left-frame menu of the main CCRMA website to learn more about computer usage and other facilities at CCRMA.

3 Textbooks

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

**Mathematics of the Discrete Fourier Transform (DFT)**\(^7\) by Julius O. Smith

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

**Introduction to Digital Filters**\(^8\) by Julius O. Smith

See \(^6\) 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 assignment
- 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

\(^6\)http://ccrma.stanford.edu/guides/
\(^7\)http://ccrma.stanford.edu/~jos/mdft/
\(^8\)http://ccrma.stanford.edu/~jos/filters/
“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
  • General in-class discussion
  • Getting to know your fellow class-members better

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 result in a good amount of learning.

6 Schedule and Pointers

Note: The online version\(^9\) of this schedule contains hyperlinks to all reading, lecture videos, and assignments.

To obtain printable versions of the assignments and solutions from off-campus locations, you can use commands such as

```
scp you@ccrma-gate.stanford.edu:/usr/ccrma/web/html/courses/320/hw/hw1x/hw1x.pdf .
scp you@ccrma-gate.stanford.edu:/usr/ccrma/web/html/courses/320/hw/hw1x/hw1xsol.pdf .
```

where you refers to your CCRMA login, and x is a for 320A and b for 320B.
You can alternatively use VPN\(^10\) (Virtual Private Network) access.
For more info, see [https://ccrma.stanford.edu/guides/remoteaccess/](https://ccrma.stanford.edu/guides/remoteaccess/).

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\(^9\) [https://ccrma.stanford.edu/~jos/intro320/Lectures_Assignments.html](https://ccrma.stanford.edu/~jos/intro320/Lectures_Assignments.html)

\(^10\) [https://uit.stanford.edu/service/vpn](https://uit.stanford.edu/service/vpn)
6.1 Section 1: Linearity and Time Invariance; Time-Domain Representations

- Reading:
  - Chapters 1 and 2 of Introduction to Digital Filters
  - 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
  - Matrix Filter Representations
  - Optionally peruse the Music 421 overheads pertaining to acyclic convolution
  - Supplementary: Audio Signal Processing in Faust
  - Assignment 1

- Lecture Videos (Total Viewing Time ≈ 2 Hours):
  IMPORTANT NOTICE: The videos are hosted on YouTube and they use annotations for corrections and supplementary information. These annotations are not supported on mobile devices. It is therefore unfortunately important to view these videos in a Web browser on a desktop/laptop computer.

  - Derivation of Convolution from Linearity and Time-Invariance (LTI) (Superposition)
  - General Linear [Causal] [Time-Invariant] Filters — Matrix Representations
  - Recursive Filters, Simplest Lowpass, Phase Delay, Group Delay
  - Supplementary: Faust in the Classroom
  - Supplementary: Faust Intro
  - Supplementary: Faust Implementation of the Simplest Lowpass Filter
  - 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

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12 https://www.youtube.com/watch?v=p19QzBxnhvg
13 https://www.youtube.com/watch?v=KWhqV95fKkw
14 https://www.youtube.com/watch?v=S7ye_H1A_hc
15 https://www.youtube.com/watch?v=r0fg6eZAKGs
16 https://www.youtube.com/watch?v=21Et7dszl00
17 https://www.youtube.com/watch?v=qE1_UzQZmnM
18 https://www.youtube.com/watch?v=jNcKGlMHE9A
19 https://www.youtube.com/watch?v=1J7mnqyVBfk
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 [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 [21] [5:31]

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 [22] [27:29]

6.2 Section 2: Transfer-Function and Pole-Zero Analysis of Digital Filters, Analog and State Variable Filters, Digitizing Filters

- **Reading:**
  - Appendix D (Laplace Transform Analysis)
  - Chapter 6 (Z-transform), Chapter 6 (Transfer Function Analysis)
  - Appendix E (Analog Filters)
  - Laplace Analysis of a Force-Driven Mass
  - Appendix I.3 (Bilinear Transform)
  - Supplementary: Digital State-Variable Filters
  - Supplementary: Interactive Möbius Transformation

- **Assignment 2**

- **Lecture Videos (Total Viewing Time \( \approx \) 3 Hours):**
  - Transfer Functions, Partial Fraction Expansion, Repeated Poles [23] [44:52]
  - Transfer Function [24] [50:43]
  - State Variable Analog Filters and Digitization [25] [47:50]
  - Repeated Poles at \( s = 0 \)
    - * One Pole at DC in the s Plane [24] [14:17]
    - * Mechanical Integrator using a Mass [27] [3:46]

20 https://www.youtube.com/watch?v=qZUcTsHkHBQ
21 https://www.youtube.com/watch?v=BULkMAst6_U
22 https://www.youtube.com/watch?v=TVJpTqHjF8c
23 https://www.youtube.com/watch?v=fRLfliem52M
24 https://www.youtube.com/watch?v=3C3K7PSxCwg
25 https://www.youtube.com/watch?v=CBpVm9fH7Hs
26 https://www.youtube.com/watch?v=DIhH2JtQeE
27 https://www.youtube.com/watch?v=c1f1X2Ybn3U
* Integrator made by a Spring or Inductor [28:5:46]
* One Pole at DC in the s Plane, Continued [23:1:31]
* Frequency Response of an Integrator [30:5:05]
* Repeated Poles at DC [51:4:16]
* General Transfer Function of a Pile of Poles at DC [53:3:38]
* Impulse Response of a Pile of Poles at DC [53:3:22]

- Simplest Electrical LPF: RC lowpass; RLC Circuits: Resistor Equation \( V = IR \), Capacitor Equation \( Q = CV \), Inductor Equation \( V = L \frac{dI}{dt} \); 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 [54:21:39]
- Bilinear Transform = special case of Moebius Transformation [DON’T MISS THIS ONE!] [50:2:34]

### 6.3 Section 3: Frequency-Response Analysis, Quality Factor \( Q \), Allpass Filters

- **Reading:**
  - Chapter 7 (Frequency Response Analysis)
  - First three sections of Chapter 8 (Pole-Zero Analysis)
  - Second section of Chapter 9 (Implementation Structures) on parallel/series filter sections
  - Appendix B (Elementary Audio Digital Filters) on one/two pole/zero sections, allpass filters, dc blockers, low and high shelf, peaking equalizers
  - Appendix C (Allpass Filters), through the first subsection (i.e., the rest is “supplemental” starting at Paraunitary Filters)
  - Supplementary: Robust Design of Very High-Order Allpass Dispersion Filters [37]
  - Review: Complex Resonators (PDF) [38]
  - Review: Comparing Analog and Digital Complex Planes [39] from last quarter

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28 https://www.youtube.com/watch?v=qPLyMge71F4
29 https://www.youtube.com/watch?v=1qXYMuna3yw
30 https://www.youtube.com/watch?v=VLtpw4eD4Uc
31 https://www.youtube.com/watch?v=5s6umF7I4T4
32 https://www.youtube.com/watch?v=6_VRX5Fvdig
33 https://www.youtube.com/watch?v=d6uQYhQteaw
34 https://www.youtube.com/watch?v=dEmmtsN-ka4
35 https://www.youtube.com/watch?v=GrpAgeVbUWs
36 https://www.youtube.com/watch?v=JX3VmgDg1FnY
• Assignment 3

• Lecture Videos

  – Simplest Electrical LPF: RC lowpass, continued; Bode Plot; 3dB Bandwidth [7:45]
  – Bandwidth of a Pole, Continuous-Time Complex Resonator and Allpass Filter, Magnitude and Phase Response from Factored Transfer Function [12:30]
  – Analog Low-Shelf Filters, High Shelf, Peaking Equalizer, Mapping s to z, Bilinear Transform (BLT), BLT Doesn’t Alias, BLT Frequency Warping [12:30]
  – Bilinear Transform Frequency Scaling, Resonance Preservation; Digitizing an Integrator (Mass), RC Filter, Low Shelf; BLT Stability Preservation, DC Blocker [8:51]
  – Supplementary: Shelf Filters in Faust [22:25]
  – Analog Filters Reviewed: Transfer Function, Frequency Response, Power Response; Analog Lowpass Design, Maximally Flat Passband, Butterworth Filters [30:46]
  – Quality Factor (Q) of a Resonator [7:12]
  – Complex One-Pole Resonator and its Q; Canonical Form of a Biquad (s-plane); Mechanical and Electrical Resonators; Limiters, Compressors, Expanders [39:30]
  – Filter Decay Time is about Q Periods [3:16]

6.4 Section 4: Linear and Minimum Phase Filters, Recursive Digital Filter Design, Butterworth Filters, More on Allpass Filters, More on State Space

• Reading:

  – Chapter 11 (Filters Preserving Phase)
  – Chapter 12 (Minimum-Phase Filters)
  – Appendix F (Matrix Filter Representations)
  – Appendix G (State Space Filters)
  – Supplementary: The Digital Waveguide Oscillator
  – State-Space Formulation of Multi-Input, Multi-Output (MIMO) Linear Filters

https://www.youtube.com/watch?v=MOBH66RyXzw
https://www.youtube.com/watch?v=m4zCmvvKFso
https://www.youtube.com/watch?v=RqYDdcDW3dY
https://www.youtube.com/watch?v=aaGdgf65PnY
https://www.youtube.com/watch?v=9RD4ylap7E
https://www.youtube.com/watch?v=doDmZfEfbg
https://www.youtube.com/watch?v=V04yxnqBuYk
https://www.youtube.com/watch?v=zeClzrK0fG0
https://www.youtube.com/watch?v=kPKQ16EdC0
https://www.youtube.com/watch?v=FEf7dpApd6I
https://www.youtube.com/watch?v=Ao1Zr1Z18nY
- Digitizing the State-Variable Filter
- Appendix I (Recursive Digital Filter Design)
- Appendix K (Digital Filtering in FAUST and Pd)

Supplementary: Audio Signal Processing in FAUST
Supplementary: State-Space Canonical Forms
Supplementary: State Variable Filter used in the ARP2500 analog synthesizer
Assignment 4

- Lecture Videos

- 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[51] [35:19]
- Butterworth Power Response, Analytic Continuation, Butterworth Poles, Matlab butter() Function[52] [7:02]
- Example Butterworth Filter of Order 2, Digitization via Bilinear Transform, Frequency Prewarping[53] [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[54] [10:30]
- 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()[55] [31:10]
- Supplementary Note on Repeated Poles[56] [4:40]
- Supplementary: MicroModeler Digital Filter Design Tools[57] [5:23]
- Minimum Phase Filters and Signals; Allpass Filters: Poles and Zeros, Graphical Amplitude and Phase Response, Biquad Realization, Phasing; Allpass-Minimum-Phase Decomposition[58] [28:44]
- Allpass Filters in z and s Planes; Instability as Noncausality; Laurent Series; Bilateral DTFT; Cepstrum; Converting Arbitrary Spectra to Minimum-Phase Form[59] [39:13]

[51]https://www.youtube.com/watch?v=pUtUrzVHF3Q
[52]https://www.youtube.com/watch?v=nhhuAxBUleU
[53]https://www.youtube.com/watch?v=UcTHnf4B5tU
[54]https://www.youtube.com/watch?v=I0g5r0_BgRM
[55]https://www.youtube.com/watch?v=MxxDS01Ea5o
[56]https://www.youtube.com/watch?v=D6_AK7mfQnQ
[57]https://www.youtube.com/watch?v=FaGslPmBY_o
[58]https://www.youtube.com/watch?v=Cj6Vjp6k7NM
[59]https://www.youtube.com/watch?v=nCwji1VeXQ44
– The State Space Formulation of Linear Systems [2016 flipped-class review]
  * Adding Feedback around the Integrator Chain, Derivation of the State Space Formulation
  * State Space Formulation, Continued
  * State Space Overview
  * Force-Driven Mass Revisited
  * General Discussion of State Space
  * Defining State Variables
  * State Variable Choice Summary
  * General State Space Model and Digitization via Backward Euler
  * State Space Converts Nth-Order to Vector First-Order
– Supplementary Review (because we already did this in Section 2): State Variable Lowpass, Bandpass, and Highpass
  * Normalized Biquad Lowpass Filter, Continuous Time
  * State Variable Realization of Normalized Biquad Lowpass
  * State Variable Filter Lowpass/Bandpass/Highpass
  * State Variable Filter LP/BP/HP Frequency Scaling and Digitization

6.5 Section 5: Cross-Overs, Moog VCF, Voice Synthesis, F0 Estimation, Cepstra, Converting to Minimum Phase

• Reading:
  – Review all assigned reading to date, slowing down where needed for full understanding

• Supplementary:
  – LinkwitzRiley filter (aka “Butterworth-squared filter”)
  – Linkwitz-Riley Crossovers: A Primer
  – Active Crossover Networks for Noncoincident Drivers in JAES Volume 24, Issue 1, pp. 2-8, February 1976
  – Spectral Envelopes via Cepstrum or LPC [from Music 421 overheads]

• Lecture Videos:
  60 https://www.youtube.com/watch?v=a__oH8rYPHc
  61 https://www.youtube.com/watch?v=ZN02687Nch0
  62 https://www.youtube.com/watch?v=Ygg66m9Nrlk
  63 https://www.youtube.com/watch?v=Zulbjgb6QQ8
  64 https://www.youtube.com/watch?v=F0ku7LwHr10
  65 https://www.youtube.com/watch?v=45OWhHpWMvU
  66 https://www.youtube.com/watch?v=LgggKFphu0o
  67 https://www.youtube.com/watch?v=rbrJkcnb_du
  68 https://www.youtube.com/watch?v=zd02nQKeaAY
  69 https://www.youtube.com/watch?v=bjHPbDsa3yk
  70 https://www.youtube.com/watch?v=9cXaE10eWuI
  71 https://www.youtube.com/watch?v=o-Zot5KoE
  72 https://www.youtube.com/watch?v=LGzSnwTFB-0
– Moog VCF[73] [23:29]
– Supplementary: Moog VCF Live-Coded in Faust
  * Moog VCF Live-Coded in Faust[74] [30:49]
  * Moog VCF in Faust, Review[75] [5:02]
  * Moog VCF in Faust, Frequency Responses[76] [7:39]
  * [Can Skip] Q-Correction and Gain-Correction Tables[77] [3:38]
– Minimum-Phase Spectra Play-List
– Complex and Real Cepstrum, Quefrency[78] [7:54]
– Mel Frequency Cepstral Coefficients (MFCC); Bark and Equivalent Rectangular Bandwidth (ERB) Psychoacoustic Frequency Scales based on Critical Bands of hearing[79] [7:54]
– Complex Cepstrum Derived; Converting Mixed-Phase Signals to Minimum Phase[80] [5:39]
– Series Expansion of Log of 1/(1-x)[81] [7:57]
– Series Expansion of Log Transfer Function in Factored Form[82] [5:26]
– Contribution of Zeros to the Complex Cepstrum[83] [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 Minimum Phase using the FFT; Testing for Time Aliasing[84] [9:14]
– Nonparametric Cepstral Folding Method in Matlab: minphasespec(), fold(), invfreqz()[85] [6:11]
– Review of another Cepstral Folding Example[86] [19:08]
– Minimum Phase Conversion by Spectral Factorization or Cepstral Method[87] [5:45]
– Minimum Phase Conversion by the Cepstral Method, Continued[88] [12:10]
– Cepstral Method Code, then CollideFX Demo by Chet Gnegy[89] [55:51]
– Supplementary: Voice Vowel Synthesis in Faust[90] [11:26]

https://www.youtube.com/watch?v=KxBbcNWbZHY
https://www.youtube.com/watch?v=WLvpGN_UN1A
https://www.youtube.com/watch?v=VVZeOCjTrc
https://www.youtube.com/watch?v=cGuuG1bp1JI
https://www.youtube.com/watch?v=QOm0njDsQYY
https://www.youtube.com/watch?v=Tlk6CLc1PPU
https://www.youtube.com/watch?v=201BBeWQ0GE
https://www.youtube.com/watch?v=OH1z4OW7UY
https://www.youtube.com/watch?v=wr1b9U7HaI
https://www.youtube.com/watch?v=x3gCA_rJ13k
https://www.youtube.com/watch?v=niY8EA4peA
https://www.youtube.com/watch?v=_yA01UTCko
https://www.youtube.com/watch?v=PTxocap8mHU
https://www.youtube.com/watch?v=V7Kr4mTM94PE
https://www.youtube.com/watch?v=7GcCkMcqYao
https://www.youtube.com/watch?v=5RNeaaFxvVg
https://www.youtube.com/watch?v=knt3bR04tI4
https://www.youtube.com/watch?v=GR97SMvS4Fw
– Supplementary: Voice Vowel Synthesis in Faust, Continued

– Supplementary: Voder [0:43]

– Supplementary: Phonem for iPad by Wolfgang Palm

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91 https://www.youtube.com/watch?v=
92 https://www.youtube.com/watch?v=e5gQBe1-z-c