DEPARTMENT OF ELECTRICAL AND COMPUTER ENGINEERING

The 1995 Mississippi State University Conference on

Digital Signal Processing

What:	EE 4773/6773 Project Presentations
Where:	Simrall Auditorium, Mississippi State University
When:	December 4, 1995 — 1:00 to 4:00 PM

SUMMARY

The Department of Electrical and Computer Engineering invites you to attend a mini-conference on Digital Signal Processing, being given by students in EE 6773 — Introduction to Digital Signal Processing. Papers will be presented on a wide range of topics including speech and image processing, parallel processing, and acoustic echo cancellation.

Students will present their semester-long projects at this conference. Each group will give a 10 minute presentation, followed by 5 minutes of discussion. After the talks, each group will be available for a live-input real-time demonstration of their project. These projects account for 50% of their course grade, so critical evaluations of the projects are welcome.

Session Overview

1:00 PM — 1:10 PM:	J. Picone, Introduction
1:15 PM — 1:30 PM:	**N. Doss**[1] and T. McMahon, “An Integrated Khoros and MPI System for the Development of Portable Parallel DSP Applications”
1:30 PM — 1:45 PM:	Y. Chen, L. Wang, A.M. Yusuf, and **H. Zhang**, “Noise Reduction in Laser Induced Breakdown Spectroscopy”
1:45 PM — 2:00 PM:	**X. Du**, W. Couvillion Jr., and M.H. Kiu, “Correction of Scan-Line Shifts of Digitized Video Images”
2:00 PM — 2:15 PM:	J. Beard, **S. Given**, and B.Y. Young, “DTMF Detection Using Goertzel’s Algorithm”
2:15 PM — 2:30 PM:	C.R. Jones, R. Seelam, M.E. Weber, **S. Wilson**, “Physical Modeling of a String Instrument”
2:30 PM — 2:45 PM:	K. Bush, A. Ganapath, **J. Trimble**, and L. Webster, “Real-Time Speech Endpoint Detection”
2:45 PM — 3:00 PM:	V. Allen, **M.E. Henderson**, E.S. Wheeler, and J. Williams, “Active Noise Cancellation in a Highly Reverberant Chamber”
3:00 PM — 4:00 PM:	Demonstrations in 414 and 434 Simrall

1. The person presenting the paper is shown in bold lettering.

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Volume I

Digital Signal Processing

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1. The person presenting the paper is shown in italic lettering.

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An Integrated Khoros and MPI System for the Development of Portable Parallel DSP Applications

by

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ABSTRACT

This project combines two public-domain paradigms to create a parallel software environment for DSP programming. MPI (Message-Passing Interface), a message passing system, is an evolving standard for parallel computing. Khoros is an integrated software environment for DSP. This purpose of this project is to describe and demonstrate a software design that exploits Khoros and MPI parallel libraries for the deployment of parallel DSP. The resulting system enables parallel DSP using the Khoros system for development and the MPI system for performance portability. Specifically, the new system provides MPI-based toolboxes containing data parallel modules and utilizes MPI as a means of communication between modules.

Building DSP Applications the Easy Way

- Development of DSP Applications
  - Many DSP applications are developed by combining and reusing well-understood libraries or modules.
  - Tools that provide a visual environment for combining DSP modules are widely available.
- Khoros
  - Provides an environment for the development of DSP applications that includes a visual programming tool called Cantata.
  - Modules are separate programs that communicate through shared files or shared memory.

Making Applications Faster - Parallel Computing

❒ Parallel Computing

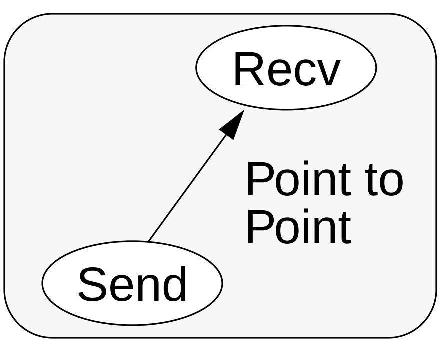
☞ More processors = Faster?

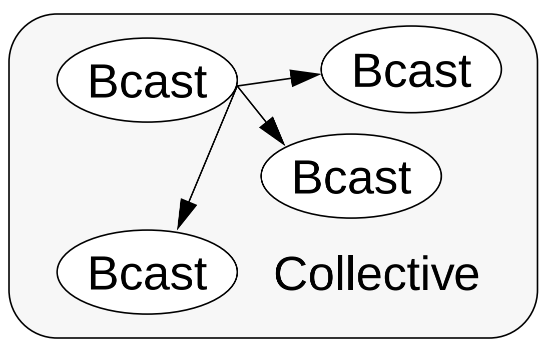
☞ Two important aspects of parallel computing:
- Speedup (benefit of additional processing resources)
- Scalability (big and fast? hardware/software)

❒ MPI (Message-Passing Interface)

☞ A standard for high-performance portable message passing.

☞ Important features
- Point to point communication
- Collective operations
- Support for parallel libraries (safety, scope)





☞ Is message passing the assembly language of parallel computing?
It's faster and scales well but in general more difficult to program with than other

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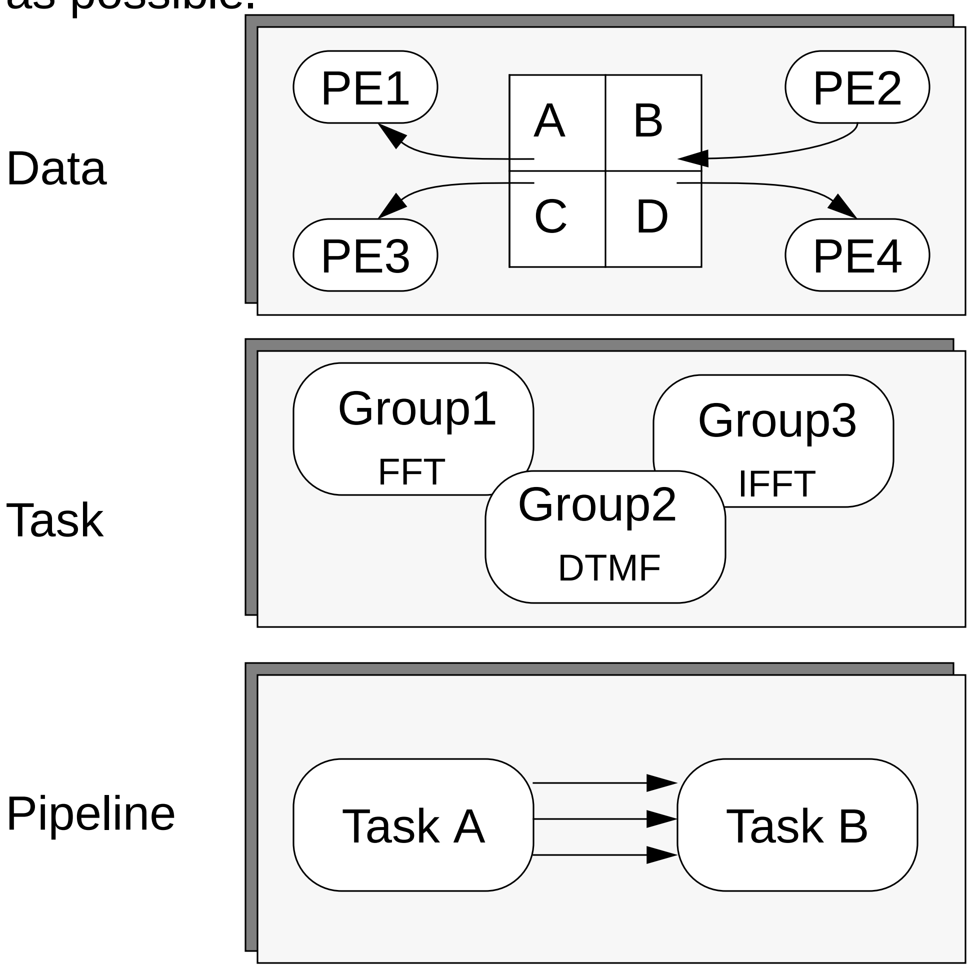
Fast, Easy Development of Portable Parallel DSP Applications?

How can we best combine a visual DSP programming environment (*i.e.* Khoros) with a message-passing programming system (*i.e.* MPI)?

- ☞ Remove burden of parallelization from user. Same interface - more knobs to tweak.

- ☞ Take advantage of as many forms of parallelism as possible.

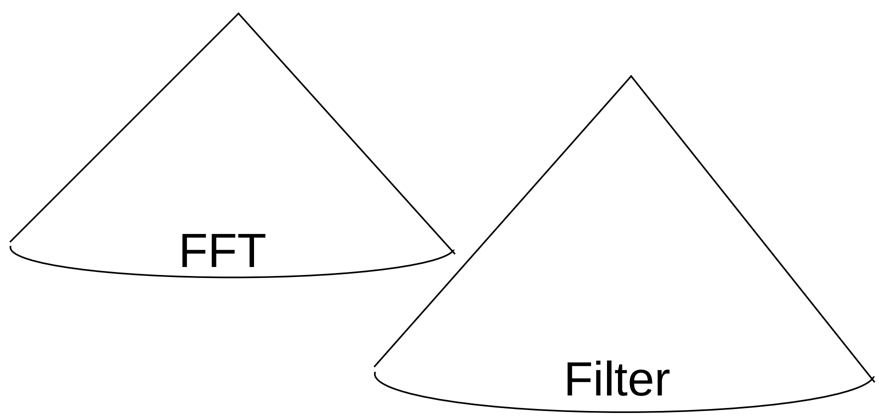


Khoros/MPI Integrated Environment
Task Parallelism

❒ MPI supports the building of independent non-interfering libraries.

Communicators
- group (scoping)
- context (safety)

Think: *Get Smart* cone of silence



❒ Khoros supports task parallelism.

Toolbox operations can execute simultaneously if there are no graph dependencies between them

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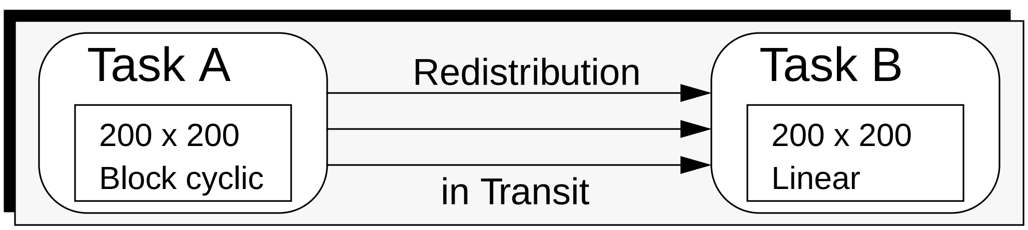
Khoros/MPI Integrated Environment
Parallel "Interbox" Communication

- Khoros uses a polymorphic data model.
  - Abstract interface to data (described by various parameters such as height, width, time, position).
  - Allows modules to operate on wide range of data.

- Data distribution independence in parallel libraries
  - An additional factor to consider in parallel libraries is data distribution (poly-algorithms).
  - Extension of polymorphic model to include data distribution



- Intergroup collective operations

  A general model for intergroup collective communication is missing in MPI

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Khoros/MPI Integrated Environment
Data Parallel Toolbox Development

- Data parallel toolboxes
  - User selects how many processes the toolbox operation will use (currently there is not an interface that allows the user to specify the machine to run on)
  - We've explored various operations for DSP in the time-domain and frequency-domain.

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Parallel Toolbox
Time-Domain

- FIR & IIR Filters
  - Direct implementation of FIR filters is embarrassingly parallel.
  - IIR filters do net lend themselves to direct parallelization.

- Convolution/Correlation
  - Parallel structures (i.e., partial fraction expansion) must be done by hand.
  - One possible approach:

Parallel Toolbox
Frequency-Domain

- DFT/FFT
  - Direct evaluation of a DFT
    k = 1 to N
    n = 1 to N
    No interdependencies between loops.
  - FFTs viewed as multidimensional (transpose algorithm)
  - Binary exchange
  - Can be viewed as matrix operations.
  - Parallel FFTs (like sequential FFTs) have been well studied.
- Linear Algebra
- Filter Generation

SUMMARY

❐ Khoros and DSP

Operations too fine-grained (flexibility is benefit) and thus must be parallelized individually.

Too much overhead (other systems compile to a single program, data model). More work should possibly be done at compile time.

The current design of Khoros is not a good fit with high-performance computing.

❐ MPI

Intercommunicator collective operations have been proposed as a part of the MPI2 standard. These type of operations fit naturally with pipeline-structured applications.

The coarse-grain message passing model is not a good fit with Khoros (as it is currently implemented).

❐ Implementation

MPI supported in Khoros (program development, process start-up).

Setup for intergroup communicators has been implemented however this has not been integrated into the Khoros polymorphic data model. A basic set of intergroup collective operations has been implemented

A few simple DSP operations have been implemented however the current implementation is far behind our original goals.

REFERENCES

1. Nathan Doss, William Gropp, Ewing Lusk, and Anthony Skjellum. A Model Implementation of MPI. Technical Report MCS-P393-1193, Mathematics and Computer Science Division, Argonne National Laboratory, Argonne, IL 60439, 1993.
2. Douglas F. Elliot. Handbook of Digital Signal Processing: Engineering Applications. Academic Press, 1987.
3. Paul M. Embree. C Algorithms for Real-Time DSP. Prentice Hall, Inc., 1995.
4. Message Passing Interface Forum. MPI: A Message-Passing Interface Standard. Technical Report Computer Science Department Technical Report CS-94-230, University of Tennessee, Knoxville, TN, May 5 1994. In the International Journal of Supercomputing Applications, Volume 8, Number 3/4, 1994.
5. Geoffrey C. Fox, Mark A. Johnson, Gregory A. Lyzenga, Steve W. Otto, John K. Salmon, and David W. Walker. Solving Problems on Concurrent Processors: Volume I, General Techniques and Regular Problems, chapter The Fast Fourier Transform, pages 187-200. Prentice Hall, 1988.
6. William Gropp, Ewing Lusk, and Anthony Skjellum. Using MPI: Portable Parallel Programming with the Message-Passing Interface. MIT Press, 1994. The example programs from the book are available from ftp://info.mcs.anl.gov/pub/mpi/using.
7. Anshul Gupta and Vipin Kumar. The Scalability of FFT on Parallel Computers. In IEEE Transactions on Parallel and Distributed Systems, volume 4, August 1993.
8. Khoral Research, Inc. Khoros Manual, 1994.
9. Vipin Kumar, Ananth Grama, Anshul Gupta, and George Karypis. Introduction to Parallel Computing: Design and Analysis of Algorithms, chapter Fast Fourier Transform, pages 377-406. Benjamin/Cummings Publishing Company, Inc., 1994.
10. Alan V. Oppenheim and Ronald W. Shafer. Digital Signal Processing. Prentice Hall, 1995.
11. John G. Proakis. Digital Signal Processing: Principles, Algorithms, and Applications. Macmillan, 2 edition, 1992.
12. Anthony Skjellum, Nathan Doss, and Kishore Viswanathan. Intercommunicator Extensions to MPI in the MPIX (MPI eXtension) Library, July 1994. Submitted to Special Issue of Integrated Computer-Aided Engineering.
13. Anthony Skjellum, Nathan E. Doss, and Purushotham V. Bangalore. Writing Libraries in MPI. In Anthony Skjellum and Donna S. Reese, editors, Proceedings of the Scalable Parallel Libraries Conference, pages 166-173. IEEE Computer Society Press, October 1993.
14. Anthony Skjellum, Nathan E. Doss, Kishore Viswanathan, Aswini Chowdappa, and Purushotham V. Bangalore. Extending the Message Passing Interface (MPI). In Anthony Skjellum and Donna S. Reese, editors, Proceedings of the Scalable Parallel Libraries Conference II. IEEE Computer Society Press, October 1994.

SPECTRUM ESTIMATION AND NOISE REDUCTION FOR LASER INDUCED BREAKDOWN SPECTROSCOPY

by

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ABSTRACT

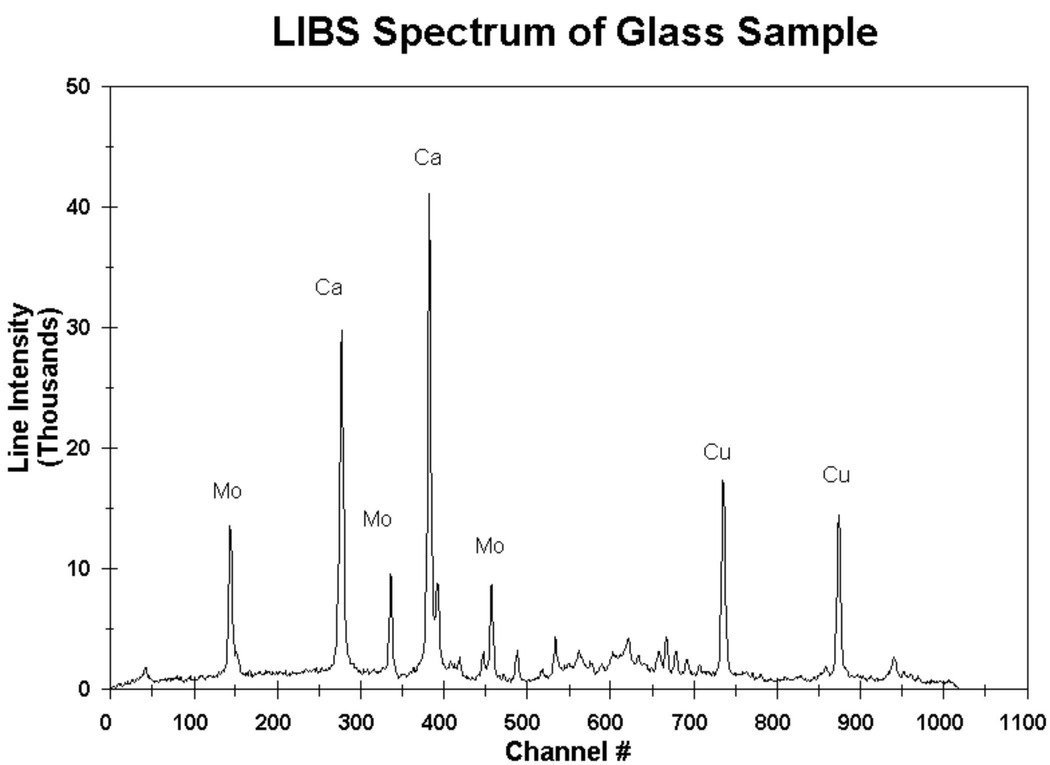
- Laser induced breakdown spectroscopy (LIBS) is a technique used to detect certain atomic and molecular species in various environment.
- Three DSP methods of autocorrelation, adaptive, and low pass filter were used to reduce the noise respectively.
- Eigenanalysis-based frequency estimation was used to estimate the LIBS spectra.

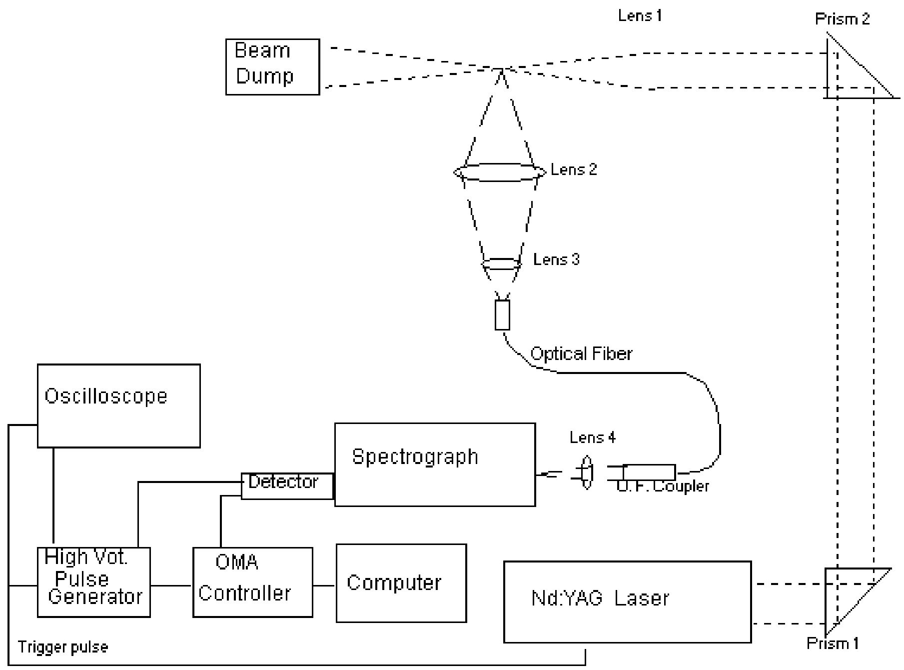
Overview of LIBS:

- A pulsed laser beam (532 nm) is focused to the target to induce a micro-plasma.
- The induced plasma produces very strong optical emission. The emission signal is collected and recorded as a spectrum with optical multichannel (1024) detector.
- Analyzes the spectrum and outputs the result.

Problem:

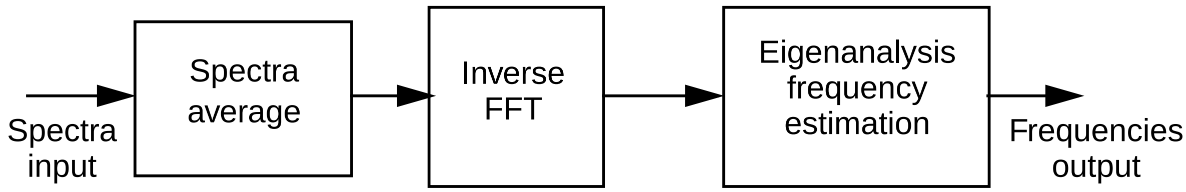
- The single spectrum is very noisy.
- The existing technology is to average 30 to 50 spectra. This limits the LIBS to perform a real time operation.




Lens 1
Prism 2
Beam Dump
Lens 2
Lens 3
Optical Fiber
Oscilloscope
Lens 4
Spectrograph
Detector
O.F. Coupler
High Vot. Pulse Generator
OMA Controller
Computer
Nd:YAG Laser
Prism 1
Trigger pulse

Algorithms:

- Eigenanalysis Frequency Estimation



- Autocorrelation:

The autocorrelation is designed to suppress the random noise corrupted signal. The signal to noise ratio can be improved by this algorithm.

$$y(n) = \sqrt{\frac{1}{L}\sum_{k=0}^{L-1} x(n)x(n+k)}$$

- Adaptive Line Enhancer:

The adaptive line enhancer is designed to suppress the noise corrupted signals. A delayed version of the input signal is used as an input to the adaptive filter whose coefficients are adapted to the best fit the portion of the input signal that is uncorrelated with the noise by minimizing the mean square error.

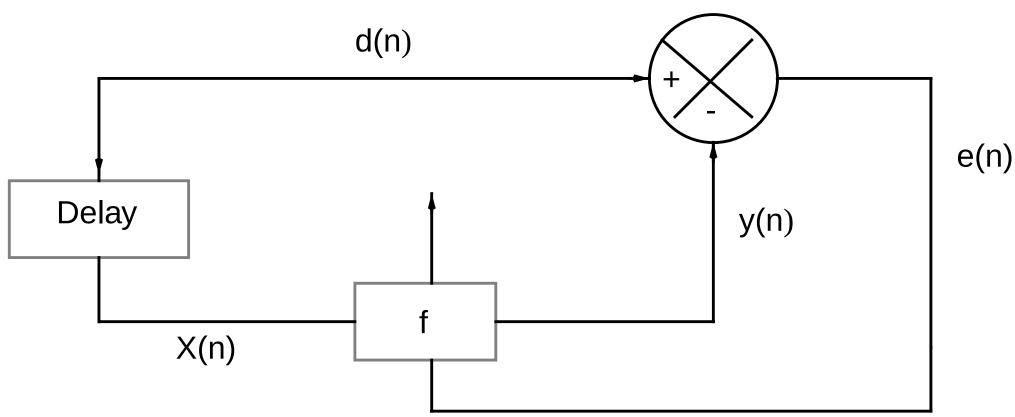


Figure 1: Adaptive Line Enhancer

Algorithms (cont.):

☞ Low Pass FIR Filter Scheme

- Low pass linear phase FIR filter in time axis

$$H_r(\omega) = h\left(\frac{M-1}{2}\right) + 2\sum_{n=0}^{(M-3)/2} h(n)\cos\left(\frac{M-1}{2} - n\right) \qquad for\ M\ odd$$

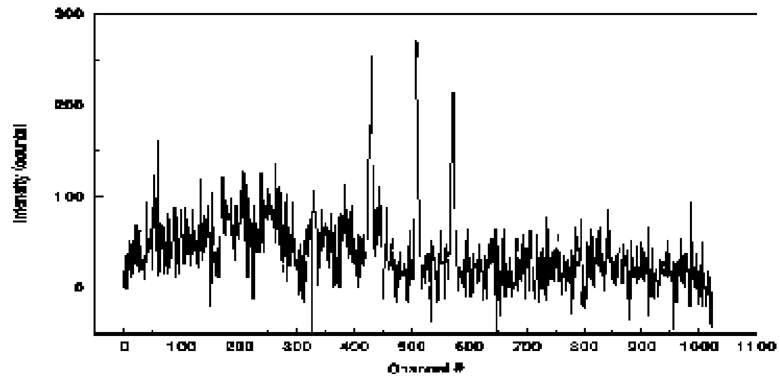
- Moving average low pass filter in frequency domain to remove background noise envelope.
- Debiasing DC component, and thresholding negative noise

☞ Evaluation and Supporting Data.

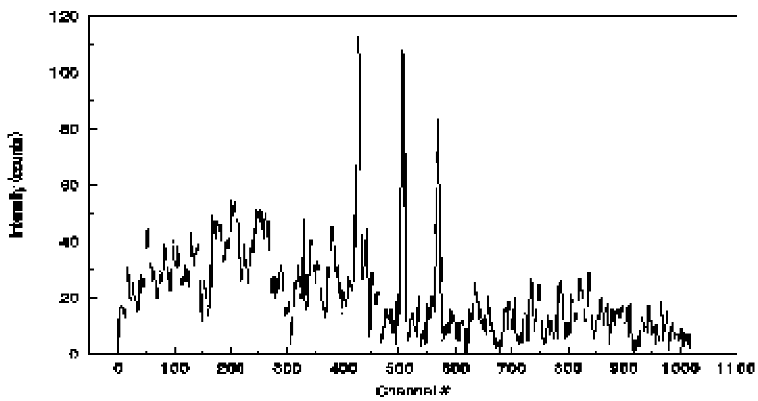
- Use mean square signal and noise ratio to evaluate the system.

$$SNR = 20\log_{10}\left(\frac{\frac{1}{NM}\sum_{i=1}^{N}\sum_{j=1}^{M} s(i,j)^2}{\frac{1}{PQ}\sum_{i=1}^{P}\sum_{j=1}^{Q} n(i,j)^2}\right)$$

LIBS Original Spectrum (set 15)

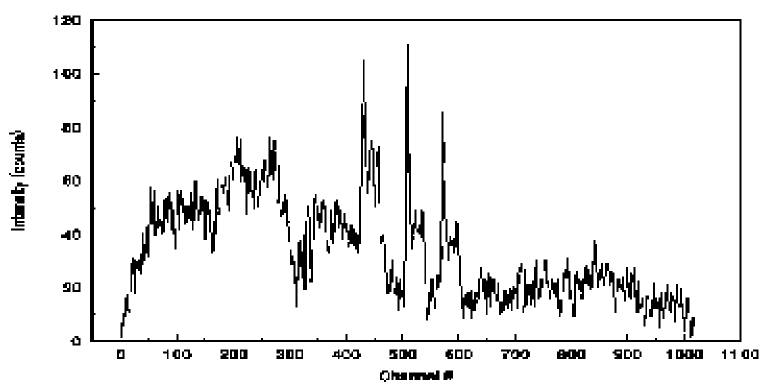


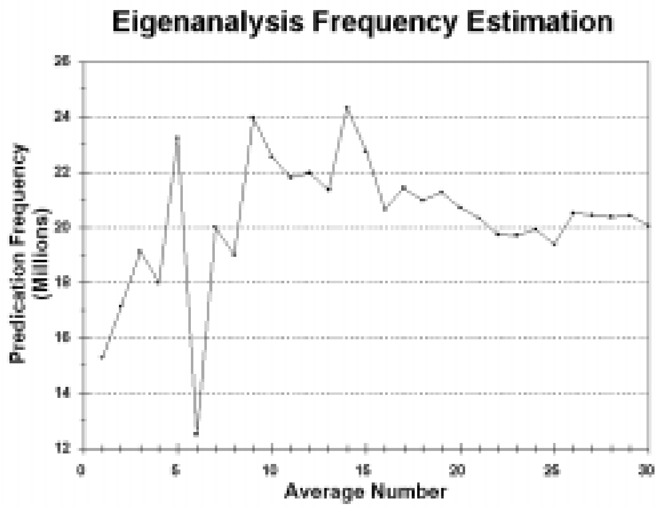
Result of Autocorrelation

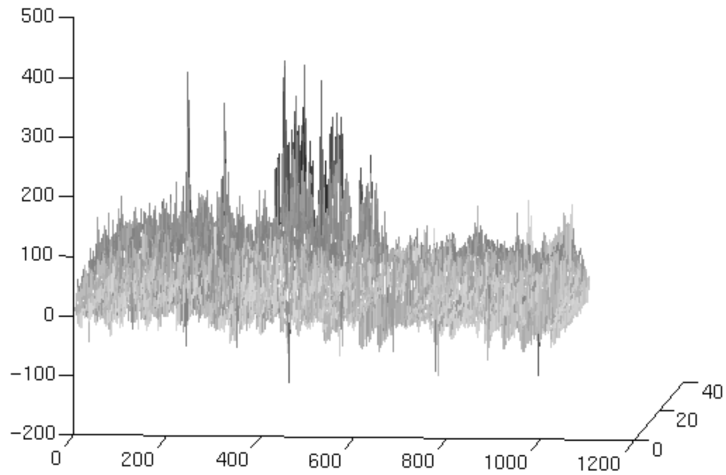


Result of Adaptive Line Enhancer

$\alpha=5\cdot10^{-7}$, L=30




Eigenanalysis Frequency Estimation
Predication Frequency (Millions)
26
24
22
20
18
16
14
12
0
5
10
15
20
25
30
Average Number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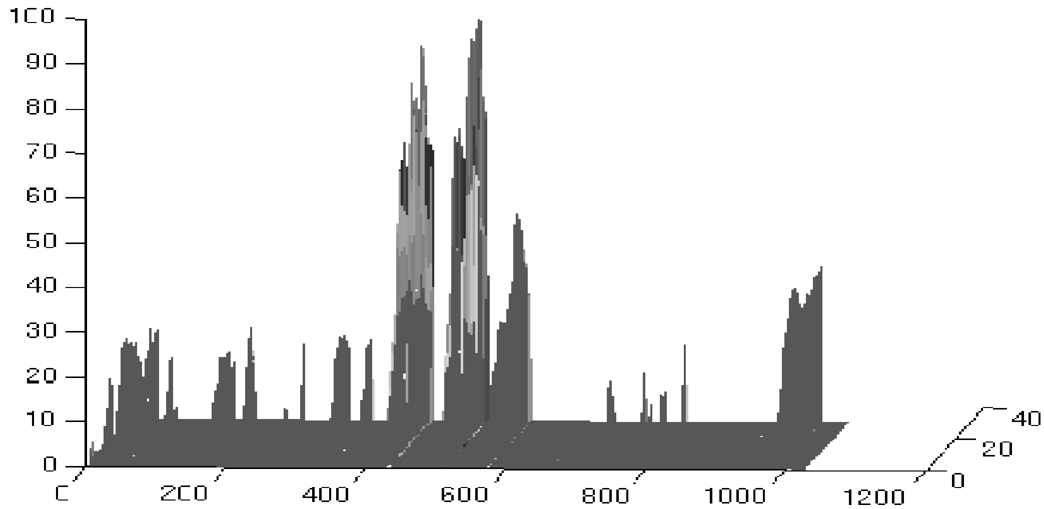
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200
400
600
800
1000
1200
0
20
40

Conclusion and Future Research:

- Our frequency estimation indicate that an accumulation of at least 20 single spectra is required to get a quality spectrum.
- SNR Improvement of DSP:

  13.1 db for low pass filter;

  4.2 db for autocorrelation;

  5.2 db for adaptive line enhancer.
- The distortion of spectral line is too serious with the method of adaptive line enhancer.
- Research on a more complicate spectra with above DSP method.
- Continue to develop new DSP technique.

Reference:

1. P. M. Clarkson, *Optimal and Adaptive Signal Processing*, CRC Press,1993.
2. S. Kay, *Modern Spectral Estimation: Theory and Application,* Prentice-Hall, NJ, 1987.
3. S. Lawrence Marple,Jr, *Digital Spectral Analysis with Applications*, Prentice-Hall, NJ, 1987.
4. B. Widrow and S. D. Stearns, *Adaptive Signal Processing*, Prentice-Hall, NJ, 1985.
5. L. Wang, N. H. Younan, and D. L. Monts, "On Using Adaptive Filtering to Extract the Absorption Cross Section of SO2 in DALAS System," in *Proceedings of the 27th IEEE Southeastern Symposium on System Theory*, 1995.

Correction of Scan-Line Shifts of Digitized Video Images

by

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The United States Forestry Service (USFS) wishes to use satellite photos to determine the amount of land covered by forests. The USFS digitized footage shot from a moving airplane using a VHS camcorder. These images were to be used as test data to test the accuracy of the satellite. The digitalization process shifted each scan-line by a random amount. Image processing techniques originally designed for detecting moving objects were modified to determine the amount each scan-line was shifted, so that a corrected image could be created by shifting the scan-lines back to their original position. These techniques approximately doubled the signal to noise ratios (SNRs) (in dB) for the special case of images with only every other scan-line shifted. Attempts to modify these techniques further to work for images where all scan-lines were shifted raised the SNR a few tenths of a dB. This was not enough to visually detect an improvement.

Introduction

- The US Forestry Services (USFS) needs to correct digitized images with each scanline shifted by a random amount.
  - ☞Images were to be used as test data for a new method of determining forest acreage using satellite.
- The USFS used inexpensive, readily available equipment.
  - ☞ A method of removing such errors would reduce the price of quality digital aerial images.
  - ☞ The method could reduce the price of creating quality digital images from VHS tapes.

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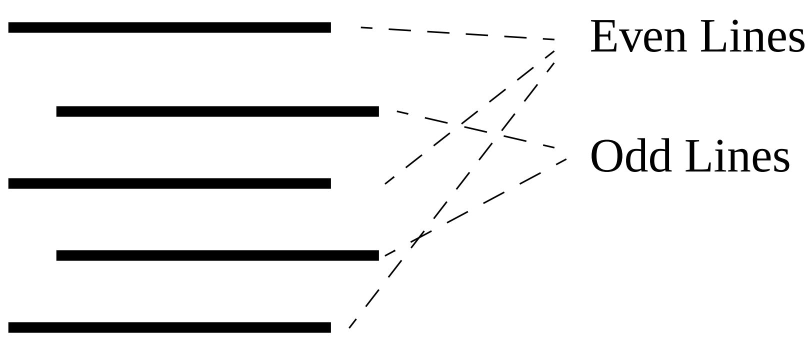
Problem Definition

❒ The images are interlaced:



❒ Source of error:

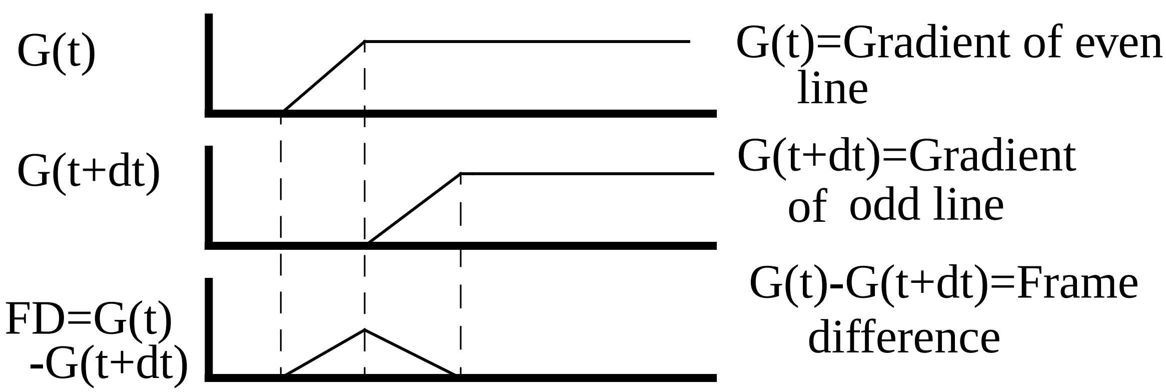
- ☞ Camera may have moved between when one set of scan-lines and the next are recorded.
- ☞ When digitizing the image from VHS tape, sync signal may not be recognizable to the digitizer.

❒ Error:

- ☞ Each scan-line is shifted, introducing "jaggies" in the images.
- ☞ Distribution of the shifts appear to be random.

Existing Technology
Pel-Recursive Algorithm

- ❒ Originally used for motion compensation of objects from 2-D images.
  - ☞ For our problem, each scan-line is treated as an object.
- ❒ Method:
  - ☞ Use Low-pass filter to remove noise from the images.
  - ☞ Edge detection.
  - ☞ Use mean filter to remove low amplitude edges from the images.



  - ☞ Shift the odd line until the “motion” or FD is minimized.

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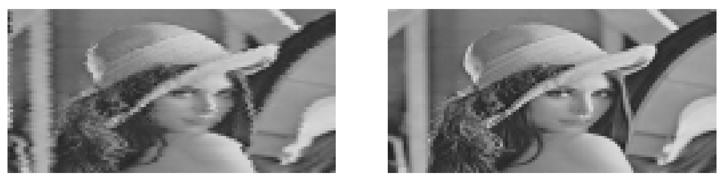
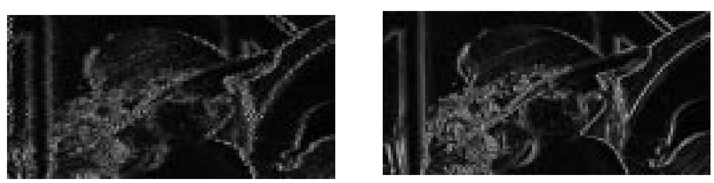
Algorithm

Generate Test Image

Split it into two images



Edge Detection



Find Shift and Compensate

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Computing Line Shifts

- Minimized absolute difference method:
  - To find the amount of shift in a line, compute:

$$Sum_i = \sum_{j=0}^{n} \left| E_j - O_{mod_n(j+i)} \right|$$

for j=0,1,...., n=number of pixels per line;
E: even line value, O: odd line value;

  - Find i for minimum sum_i.
  - Shift even line i pixels.
- Minimized square difference method: Same as above, except:

$$Sum_i = \sum_{j=0}^{n} \left(E_j - O_{mod_n(j+i)} \right)^2$$

- Circular cross correlation method:
  - Compute circular cross correlation function for odd and even lines:

$$correlation(i) = \sum_{j=0}^{n} E_j \cdot O_{mod_n(j+i)}$$

  - Find i for maximum correlation(i).
  - Shift even line i pixels.

How to determine the shifts

- Minimized absolute difference.
- Example:
  - Initial position:
    Even line: a, b, c, d.
    Odd line: q, r, s, t.
    Calculate: Sum0 = |a-q| + |b-r| + |c-s| + |d-t|
  - Shift to left by one pixel:
    Even line: a, b, c, d.
    Odd line: r, s, t, q.
    Calculate: Sum1 = |a-r| + |b-s| + |c-t| + |d-q|
  - Shift to right by one pixel:
    Even line: a, b, c, d.
    Odd line: t, q, r, s.
    Calculate: Sum2 = |a-t| + |b-q| + |c-r| + |d-s|
  - Do the above two steps iteratively for some range (for example: -10 to 10)
  - You will have a series of Sums and find min{Sum0,Sum1,....,SumN}

Why we can use max correlation's position as the shift number?

- Cauchy-Schwarz inequality:

$$\sum_{i=0}^{n} a_i \cdot b_i \leq \sqrt{\sum_{i=0}^{n} a_i} \cdot \sqrt{\sum_{i=0}^{n} b_i}$$

- If we have a,b,c,d and b,c,d,a, then:

  ab + bc + cd + da <= $a^2 + b^2 + c^2 + d^2$ and

  ac + bd + ca + db <= $a^2 + b^2 + c^2 + d^2$ etc.

- Let

  Even line: a, b, c, d

  Odd line: b, c, d, a

  Then Rx,y(0) = ab + bc + cd + da

- If we calculate Rx,y(1) = $a^2 + b^2 + c^2 + d^2$

  At this position, Rx,y(1) has the maximum value, odd line and even line are matched.

Evaluation

- Average absolute difference ratio:

$$Error = \frac{1}{xy}\sum_{0,0}^{x,y}\frac{|f(i,j) - f''(i,j)}{f(i,j)}$$

- Signal to Noise Ratio (SNR):

$$SNR = 10 \cdot \log \frac{(\Sigma f(i,j)^2)}{\Sigma(f(i,j) - f''(i,j))^2}$$

- f(i,j) = Intensity value at pixel (i,j) of the original or good image.
- f”(i,j) = Intensity value at pixel (i,j) of the corrected image.

Results

- "Lena"
  - Average absolute difference ratio = 0.091828
  - Signal to Noise Ratio = 15.86 dB

Shift Estimation Method	Average Absolute Difference Ratio	Signal to Noise Ratio
Absolute Difference	0.007375	30.03
Difference Squared	0.006254	30.81
Correlation	0.006254	30.81

- Landscape Photo1:
  - Average absolute difference ratio = 0.124022
  - Signal to Noise Ratio = 12.94 dB

Shift Estimation Method	Average Absolute Difference Ratio	Signal to Noise Ratio
Absolute Difference	0.006883	25.50
Difference Squared	0.007605	25.13
Correlation	0.007605	25.13

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Results(cont.)

- Lanscape Photo2:
  - ☞ Average absolute difference ratio = 0.100883
  - ☞ Signal to Noise Ratio = 13.04 dB

Shift Estimation Method	Average Absolute Difference Ratio	Signal to Noise Ratio
Absolute Difference	0.002321	32.50
Difference Squared	0.002717	31.77
Correlation	0.002717	31.77

- Landscape Photo3:
  - ☞Average absolute difference ratio = 0.171154
  - ☞Signal to Noise Ratio = 10.54 dB

Shift Estimation Method	Average Absolute Difference Ratio	Signal to Noise Ratio
Absolute Difference	0.009793	24.60
Difference Squared	0.011545	24.06
Correlation	0.011545	24.06

Results(cont.)

Original

Distorted,
max shift, 20 pixels

Corrected by cross-correlation method

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Results(cont.)

Distorted,
max shift,
20 pixels

Corrected by
cross-
correlation
method

Summary

❒ Scan line shifts are best estimated by:

☞Shifting until the square of the error between two lines are minimized.

☞Computing the correlation function between lines and shift the distance corresponding to the maximum value.

Future Research

❒ Find a method that works when all scan lines are shifted:

☞One approach is to find two successive frames which are similar to each other, then compensate these two frames first. After that, compensate within one frame.

❒ Grow trees in bar codes for easier forest coverage estimation.

References

1. S.C.Saxena,"*Correcting Interfield Shifts in Interlaced Imagery,*" Unpublished report submitted to Dr. Rober Moorhead, Dall, 94.
2. J.D.Foley, A.v.Dam, S.K.Feiner, J.F.Hughts, *Computer Graphics: Principles and Practice, Second Edition in C,* Addison-Wesley Publishing Company, Inc., Reading , Massachusetts, USA, 1994.
3. R.J.Clarke, *Digital Compression of Still images and Video*, Harcort Brace & Company, London, UK, 1995.
4. Gonzelez, Woods, *Digital Image Processing*, 1992.

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REAL-TIME DIGITAL DTMF DETECTION USING GOERTZEL'S ALGORITHM

by

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ABSTRACT

We present a new type of digital Dual-Tone Multifrequency(DTMF) detection scheme based on the Goertzel DFT algorithm. This detection scheme is more robust and cost-effective than conventional analog detection techniques. This algorithm is designed to provide optimal performance and exceed BellCore[1] specifications for DTMF detection. The problems associated with closely spaced signal frequencies and short tone duration are overcome by proper window and frame selection. Adaptive thresholding is provided to minimize false outputs due to noise and speech.

The algorithm was tested with a variety of data including speech, music, and DTMF tones. We find that this detector is efficient, reliable, and exceeds BellCore standards.

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DIGITAL DTMF DETECTION

❒ What is DTMF Detection?

☞ Dual-Tone Multi-Frequency

☞ DTMF: the signaling method for touch-tone.

	HIGH			
LOW	1209	1336	1477	1633
697	1	2	3	A
770	4	5	6	B
852	7	8	9	C
941	*	0	#	D

Table 1: DTMF Frequencies and Character Assignments

❒ A good DTMF detector will be robust to noise, speech, and dial tone.

SIGNAL PROCESSING CHALLENGES

- ❐ Frequencies are closely spaced
- ❐ High levels of Speech and Noise
- ❐ Variations in DTMF signals
- ❐ Real-time operation

BellCore Specifications

- ❐ Code Validity Check(one freq. / group)
- ❐ Signal Power Level
  - ☞ Adaptive Sensitivity(>9dB)
  - ☞ Twist(+4 to -8 dB)
- ❐ Timing
  - ☞ Signal Duration(min. 40 ms, reject < 24 ms)
  - ☞ Interdigit Time(> 40ms)
  - ☞ Cycle Time(> 93 ms)
- ❐ Bandwidth
  - ☞ Frequency Accept and Reject Bands(1.5 %-3.5%)
  - ☞ Windows

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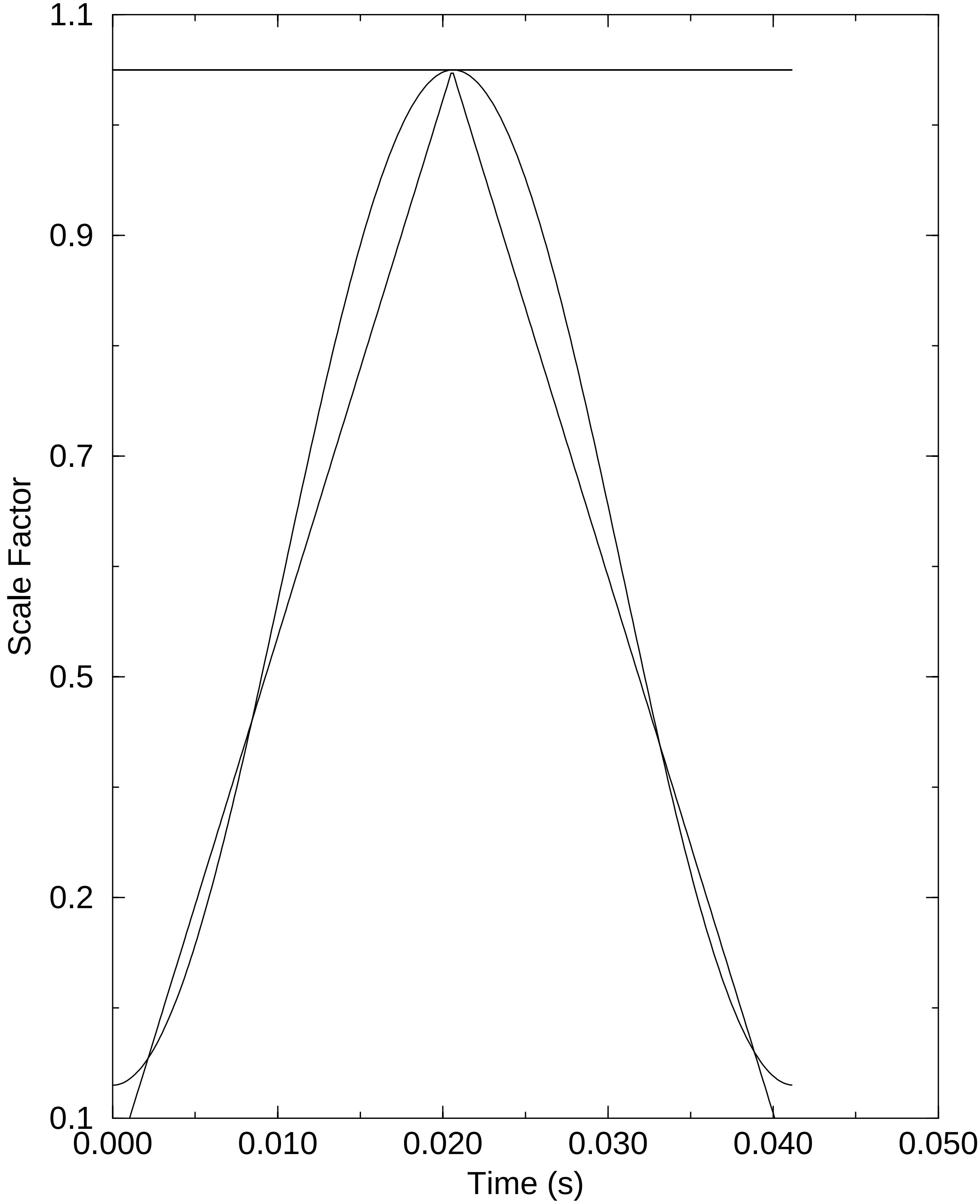
Time Windows



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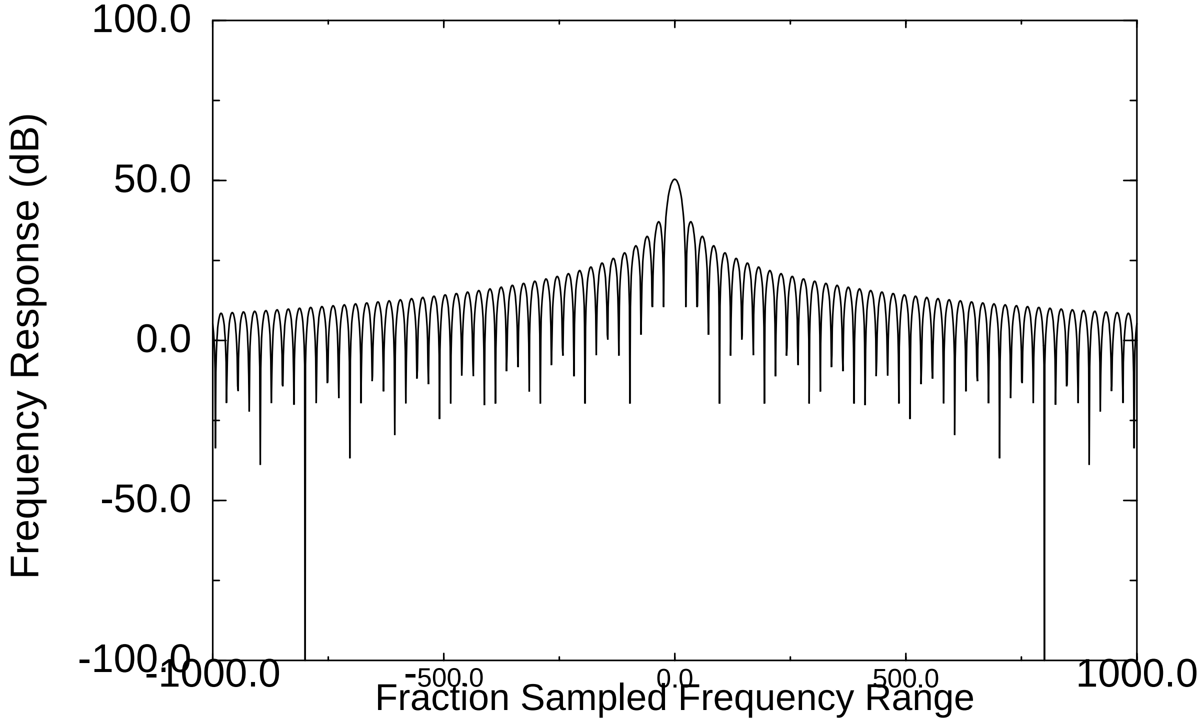
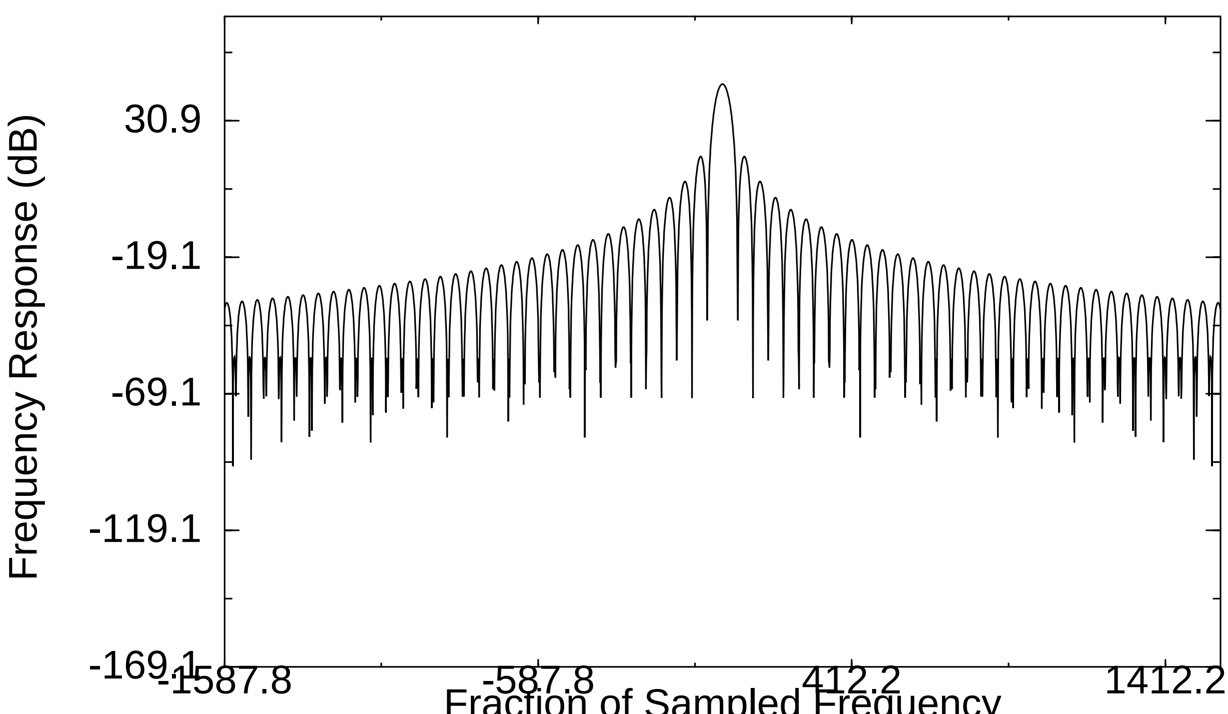
Rectangular Frequency Window



Triangle Frequency Window



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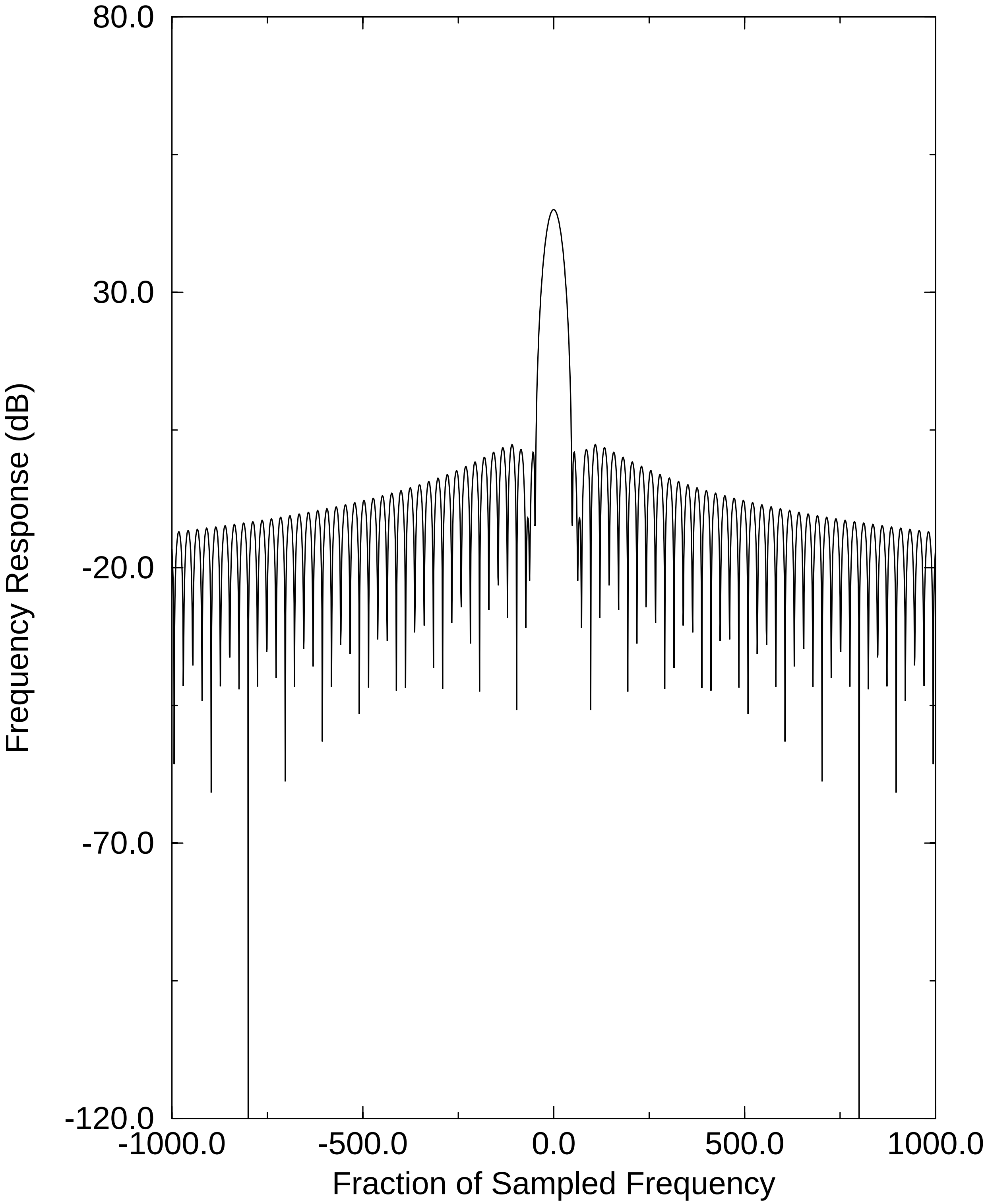
Hamming Frequency Window



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MATHEMATICAL BACKGROUND

Discrete Fourier Transform

$$\sum_{n=0}^{N-1} x(n) e^{-j2\pi k\frac{n}{N}}, k = 0, 1, \ldots, N-1$$

Rectangular Time Window

$$w(n) = 1$$

Triangular Time Window (Bartlett)

$$w(n) = \begin{cases} 2\dfrac{n}{N-1}, & n < \dfrac{N}{2} \\ 2 - 2\dfrac{n}{N-1}, & n > \dfrac{N}{2} \end{cases}$$

Hamming Time Window (Raised Cosine)

$$w(n) = 0.54 + 0.46\cos\left(\omega\frac{n}{N}\right)$$

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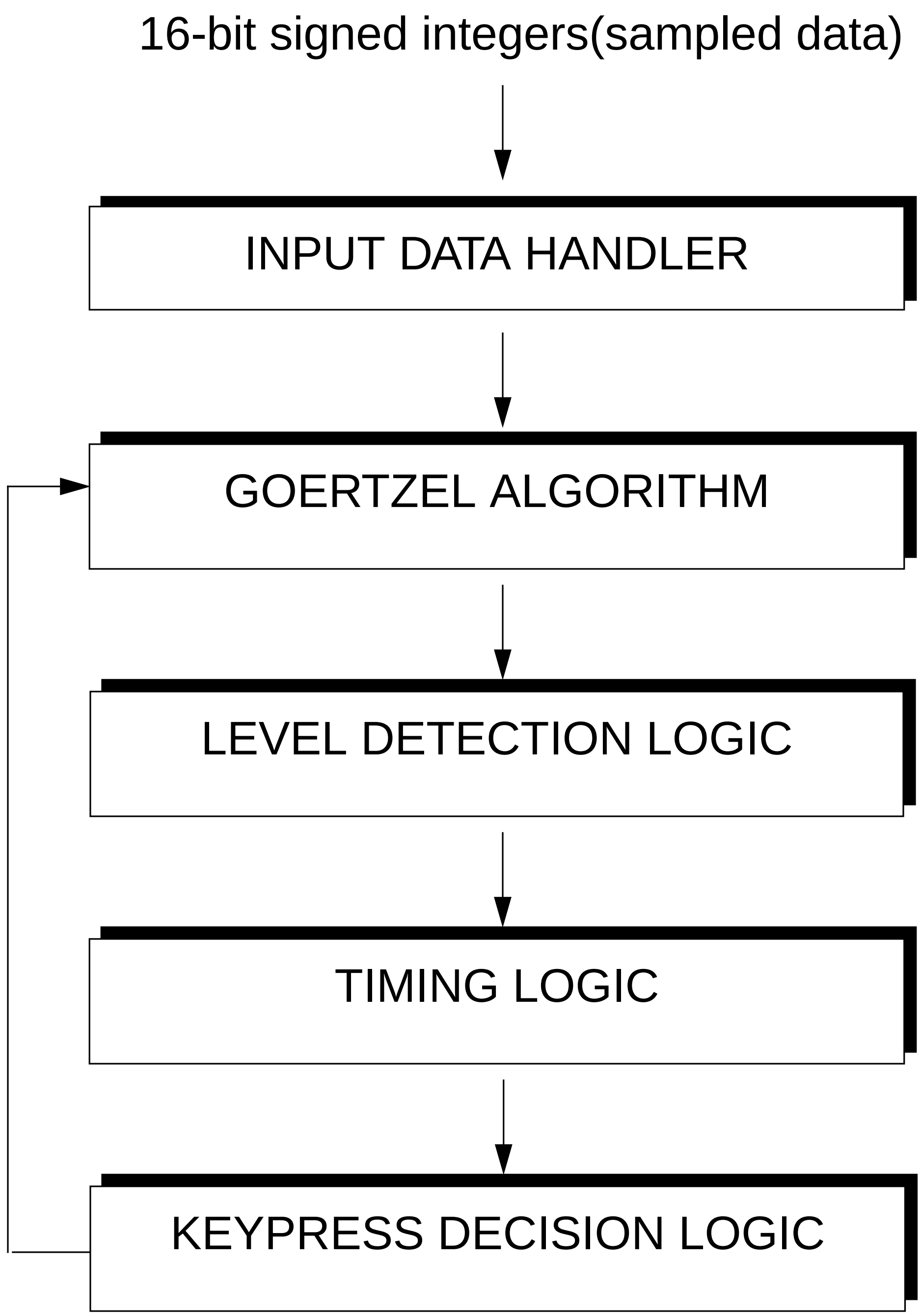
Digital DTMF Detection Based on the Discrete Fourier Transform(DFT)



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METHOD OF EVALUATION

- Computer Generated(ideal) Data
  - Tested accuracy with clean data
  - Tested accuracy with noisy data(SNR)
- Real Data
  - Tested accuracy with “clean” data
  - Tested accuracy with noisy data
  - Tested accuracy with speech and music (talk off)
  - Tested accuracy with dial tone
- Real-Time(narecord)

TESTING DATABASE

Software Generated Data	100 files 7 windows 100 files with noise added 7 windows
Recorded Data	Phone 1 3 files(92 key presses) Phone 2 2 files(124 key presses) Phone 3 1 file(99 key presses)
Speech Data	151 files(0 key presses)
Music and Speech Data	25 files(0 key presses)

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RESULTS

- No failures on talkoff
- No failures on dialtone
- No failures on music or speech
- Error free in all cases.

	Windows		
SNR	Rectangle	Triangle	Hamming
30	0	0	0
23	0	0	0
20	0	0	0
10	0	0	0
0	7	11	11
-10	84.5	89.7	92.2
-20	100	100	100
-30	100	100	100

Table 1: Percent Error for different SNR's and Windows

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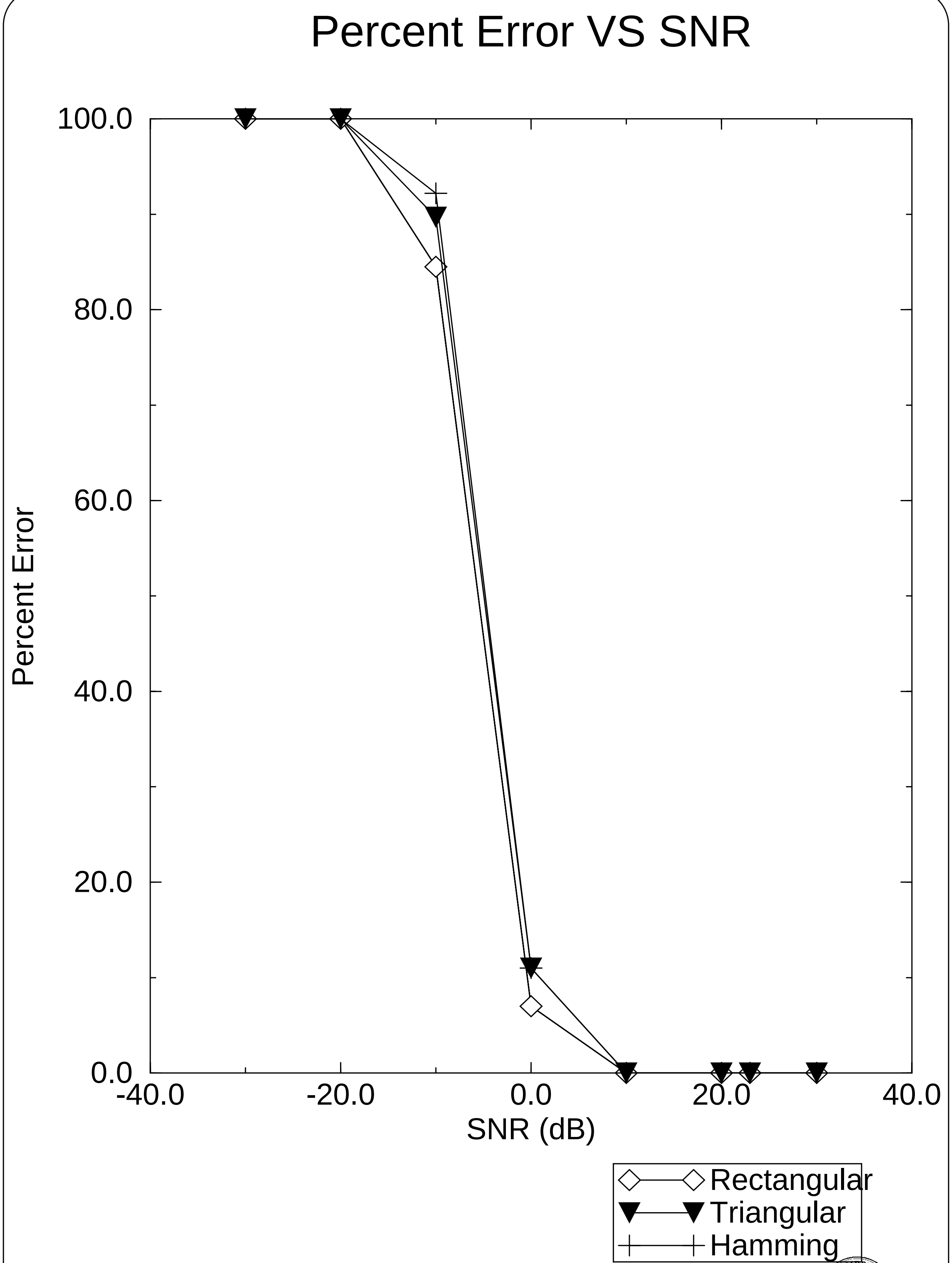

SUMMARY

- Our DTMF Detection scheme is robust to noise, speech, music, and dial tone.
- It exceeds BellCore industry standard specifications in all cases.
- Our implementation meets or exceeds performance levels of current products(i.e. SNR, DFT points)
- Our DTMF detector possesses real-time efficiency.

REFERENCES

1. Digital Simulation Test Tape." Bell Communication Research Technical Reference TR-TSY-D00763, Issue 1, July 1987.
2. Blahut, Richard E. *Fast Algorithms For Digital Signal Processing.* Addison-Wesley Publishing Co., Inc., 1985, pp. 131-2.
3. Manolakis, Dimitris G. and John G. Proakis. *Digital Signal Processing.* 2nd edition. Macmillan Publishing Co., 1992.
4. "Dual-Tone Multifrequency Receiver Generic Requirements for End-to-End Signaling Over Tandem-Switched Voice Links." Bell Communications Research Technical Reference TR-TSY-000181, Issue 1, March 1987.
5. Burrus, C. S. and T. W. Parks. DFT/FFT and Convolution Algorithms: Theory and Implementations. Texas Instruments, Inc., 1985, pp. 32-36.
6. Marple, S. Lawrence Jr. *Digital Spectral Analysis.* Prentice-Hall, Inc., 1987, pp. 136-44.
7. Keiser, Bernhard E. and Strange, Eugene. *Digital Telephony and Network Integration.* Van Nostrand Reinhold Company, 1985, pp. 289-90, 306-7.
8. Freeman, Roger L. Telecommunication System Engineering: Analog and Digital Network Design. John Wiley & Sons, 1980, 377-8.

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THE PHYSICAL MODELING OF A STRINGED INSTRUMENT

by

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ABSTRACT

TODAY:

Simplicity (Plucking a String) → **Complexity** (Building the Physical Instrument)

Complexity (Modeling a String) → **Simplicity** (Software widely Available)

FUTURE:

Complexity
(Building the Physical Instrument)

Simplicity
(Software widely Available)

OVERVIEW:

Why music synthesis?

- Saves you from hiring a professional musician
- Instrument availability for the performer
- Greater understanding of physical parameters

On the path to the future:

- Need control over physical parameters of instrument
- Physical Modeling - Wave Equation
- Digital Waveguide

Why digital waveguide?

- real time application
- model can be broken into functional blocks

What to expect?

general synthesis engine

- interchangeable models
- stand alone synth with GUI
- implementation into existing platforms

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WHAT TO IMPLEMENT

Tackling the Problem:

- A way to digitally implement a model of a stringed instrument that gives the performer expressive control.

Can this be Implemented:

- In real time?
- What are the parameters?
- Accessibility of parameters to allow expressive control?
- Limitations of the fidelity?

In Real Time?

- For any synthesizer to be useful, you must be able to "play it" and interact with it in real time.

Which Parameters to Allow Control?

- Parameters that affect performance:
  - Bow Pressure
  - Fingering Style

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WHY REINVENT THE WHEEL?

- **Existing Synthesis Technology:**
  - Fourier
    - Frequency Analysis
  - Current Synthesizers
    - Sample Based
- **All Use Static Samples**
  - One Instrument
  - One Playing Style
  - One Person
  - One Note
- **From static samples and they emulate other styles through filtering, pitch transposition, which produce sonic artifacts.**

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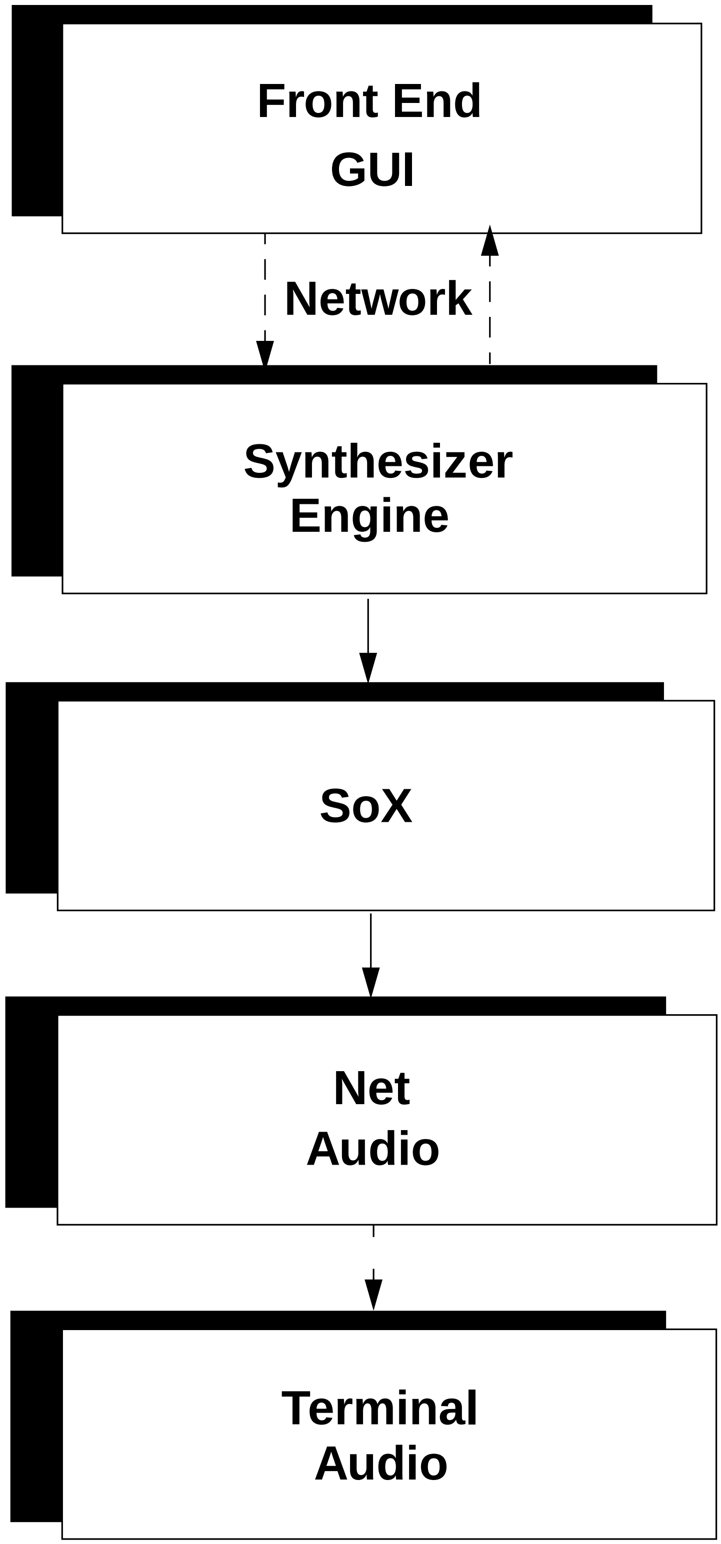
Current working Demonstration



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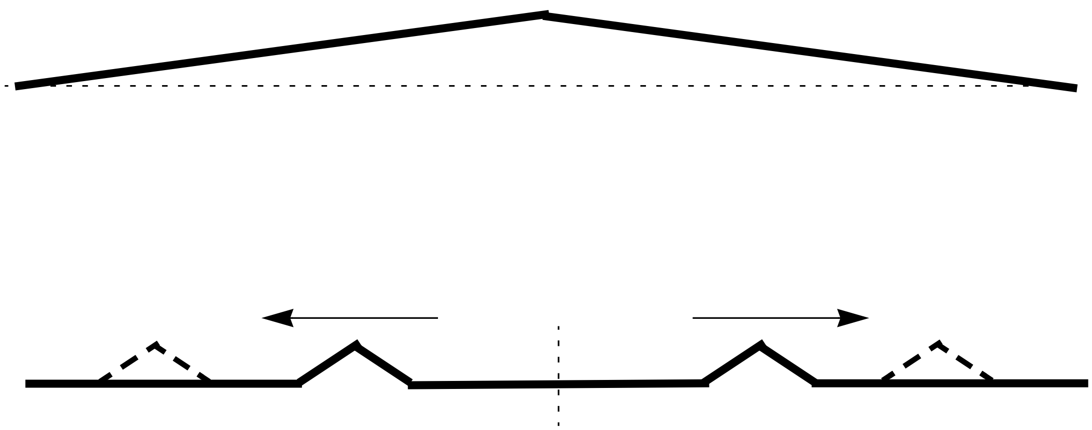
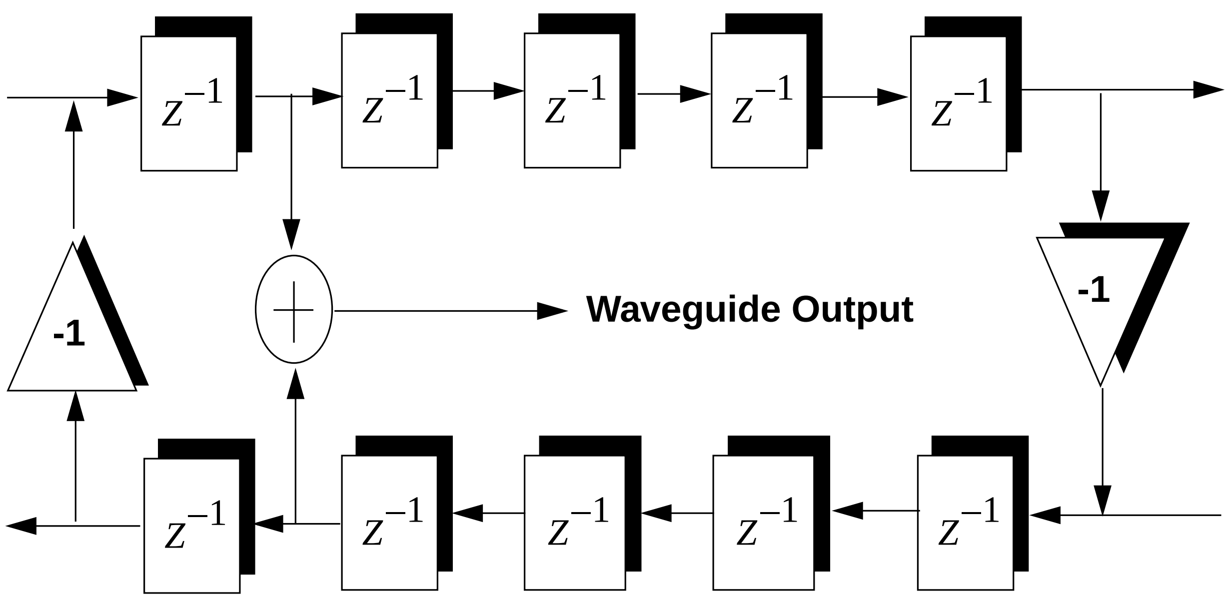
THE BREAKDOWN OF THE DIGITAL WAVEGUIDE



- **Behavior of the Plucked String**



- **Sum the Top and the Bottom Rail for the Output**

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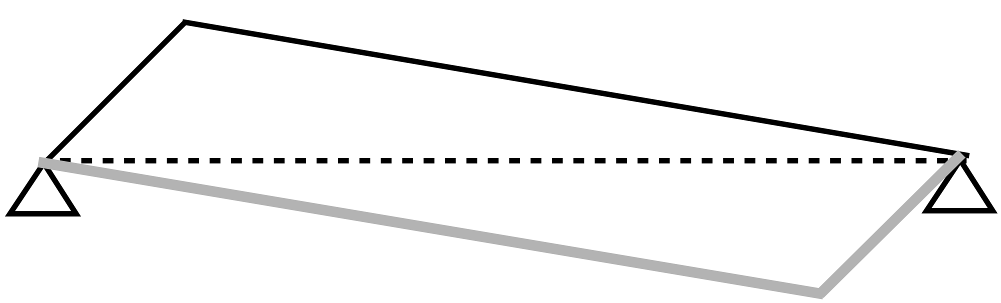
ONE DIMENSIONAL WAVE EQUATION

$$c^2 \frac{\partial^2 U}{\partial x^2} = \frac{\partial^2 U}{\partial x^2}$$

U(x,t) = displacement
c = speed of propagation

D' Alembert's Solution

$$U(x,t) = U_{pos}(x - ct) + U_{neg}(x + ct)$$



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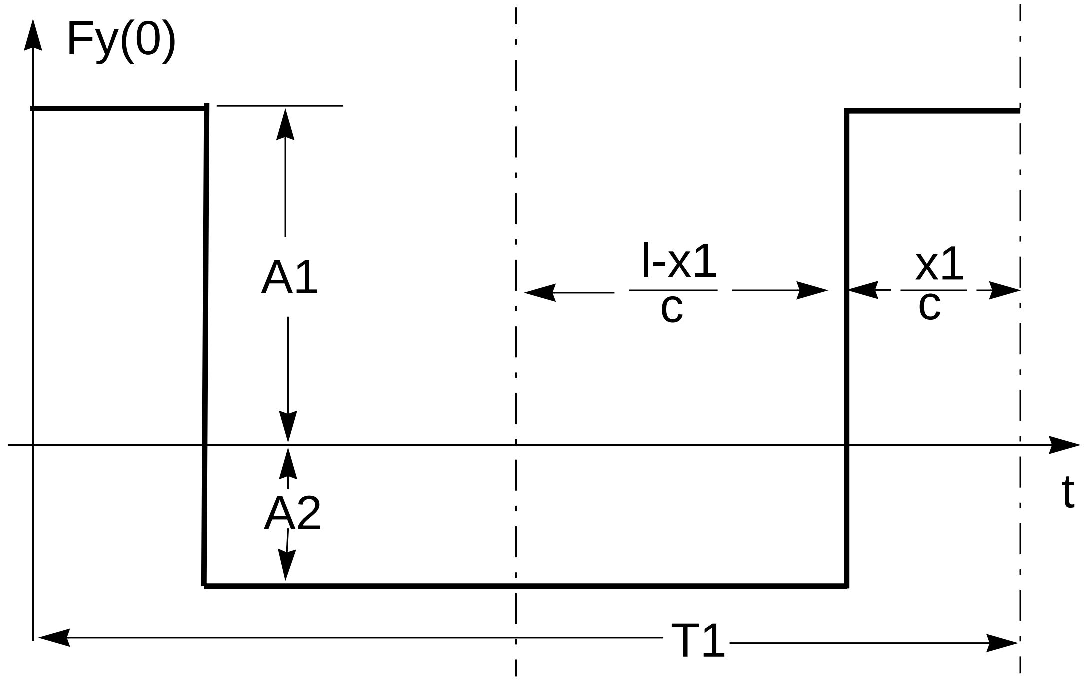
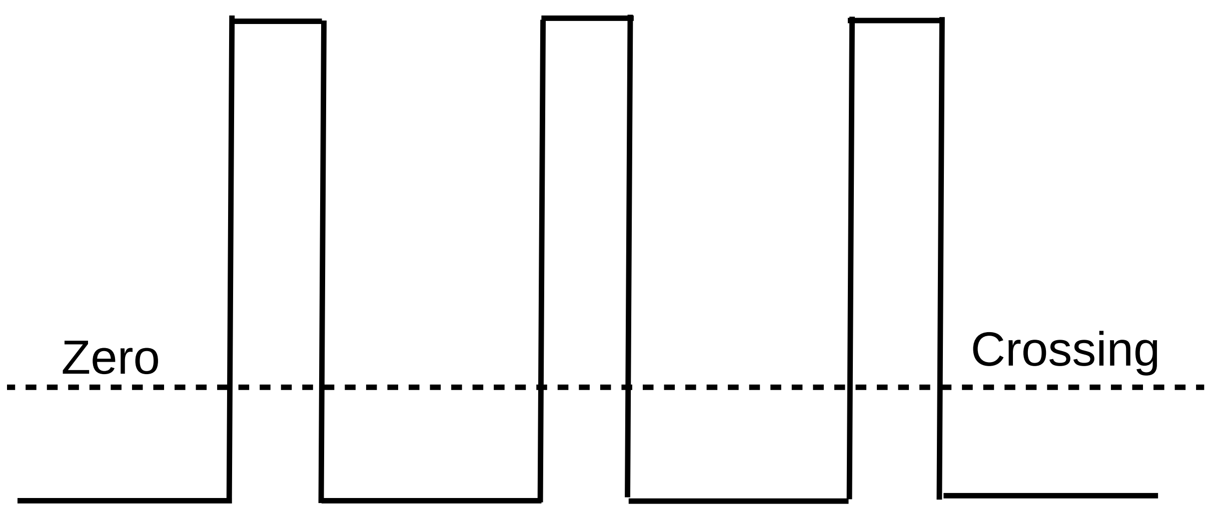


Force Function of the Plucked String



D' Alembert's Waveform of the String Plucked Near the Bridge

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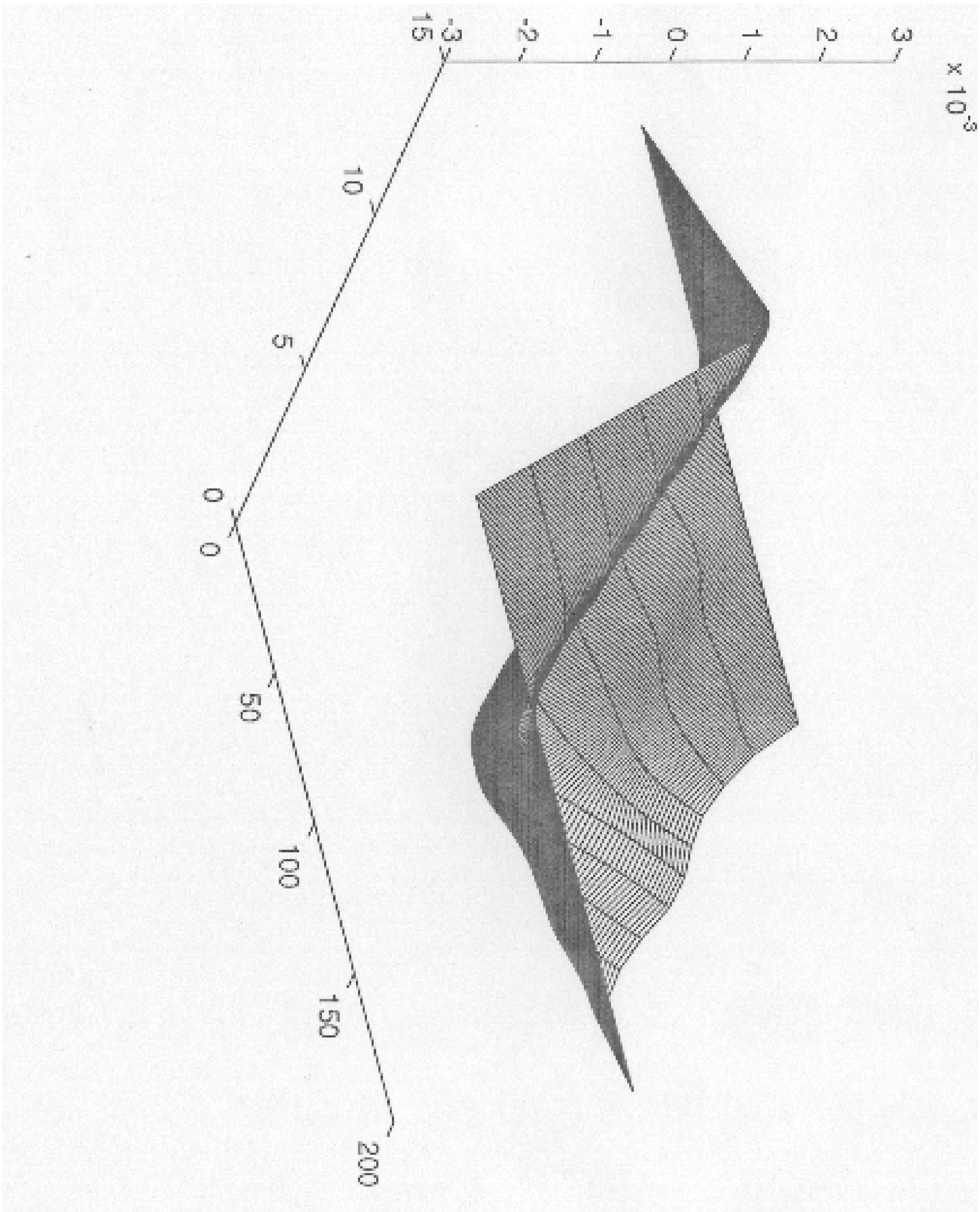

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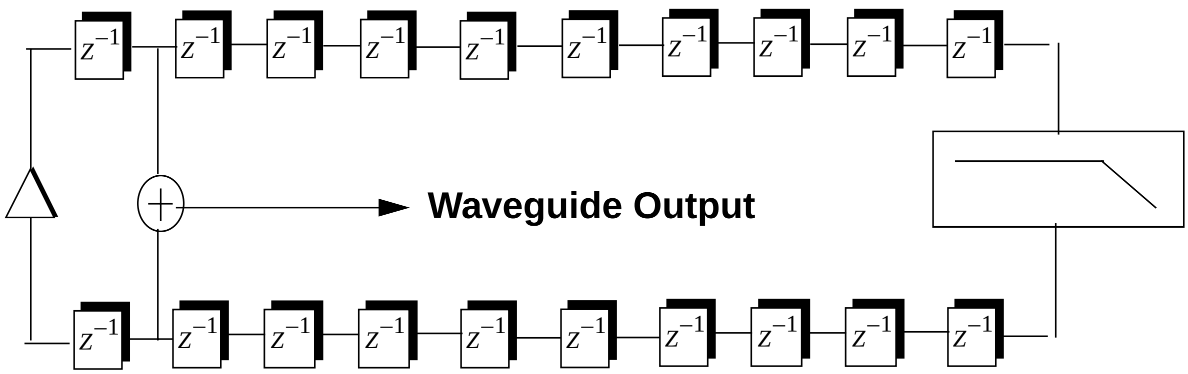
BUILDING THE DIGITAL WAVEGUIDE



- The initial value of the displacement of each point is determined by amount & location of initial displacement.
- The solution can be represented by dividing the amplitude of the initial displacement in two halves and shifting the resulting triangles to the right and left, summing amplitudes at all points.
- The end of the string supported by the nut is considered to be rigid and produces inverted reflection.
- The end of supported by the bridge can not be considered to be rigid. The bridge can be modeled with a low pass filter.
- The output can be found at any location along the string by summing the values of the upper and

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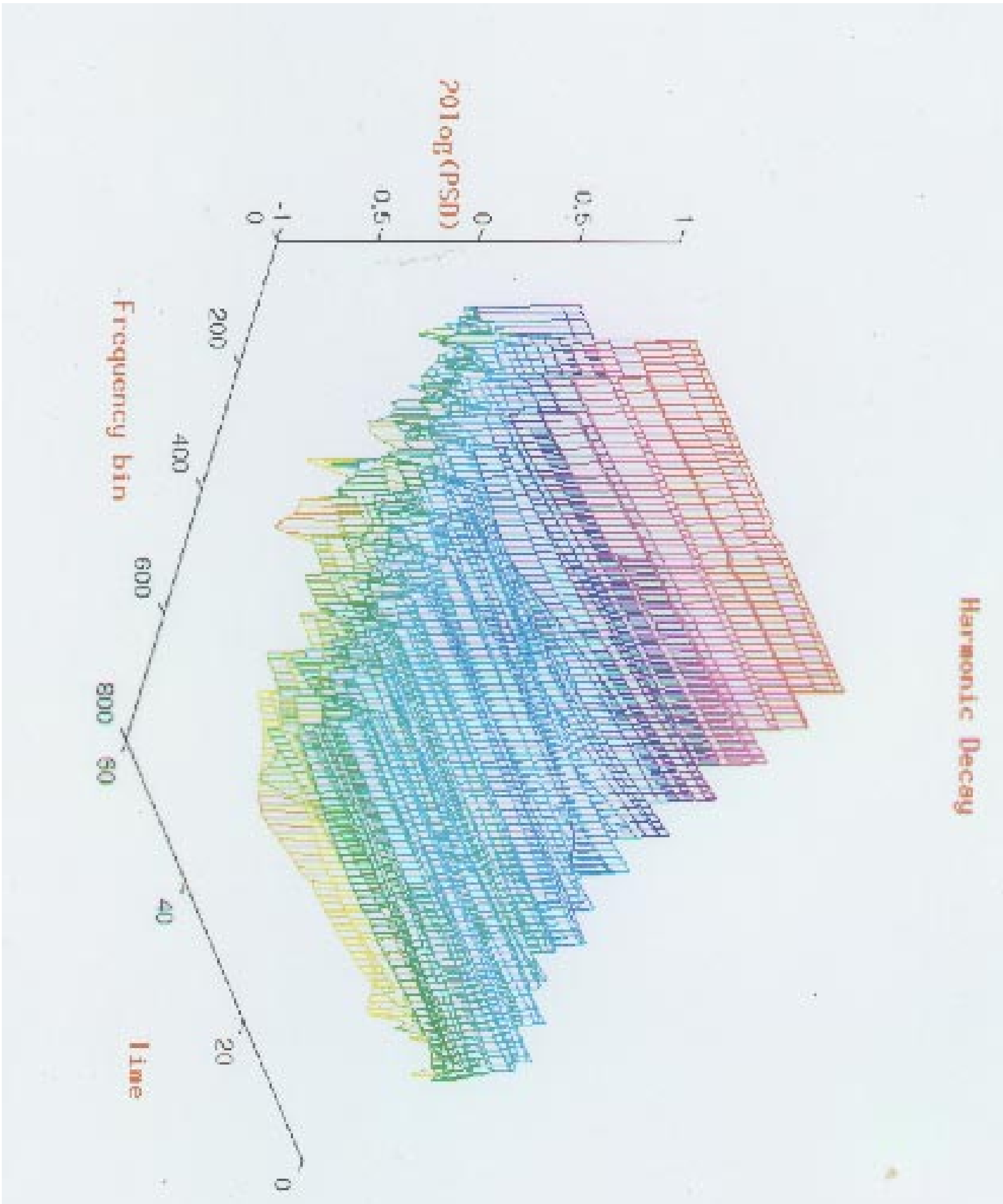
WHAT ARE WE LOOKING FOR?

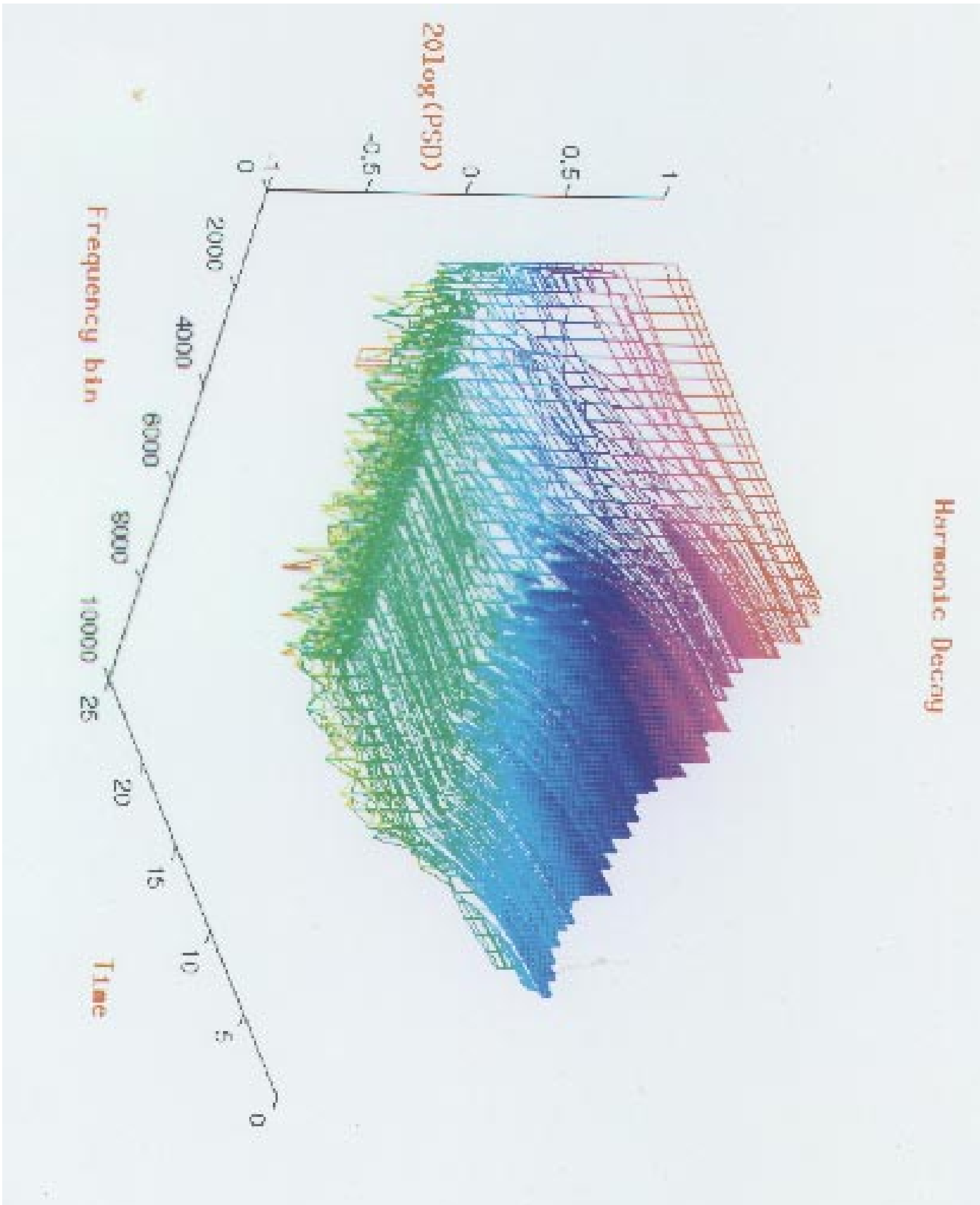
Objective Measurements:

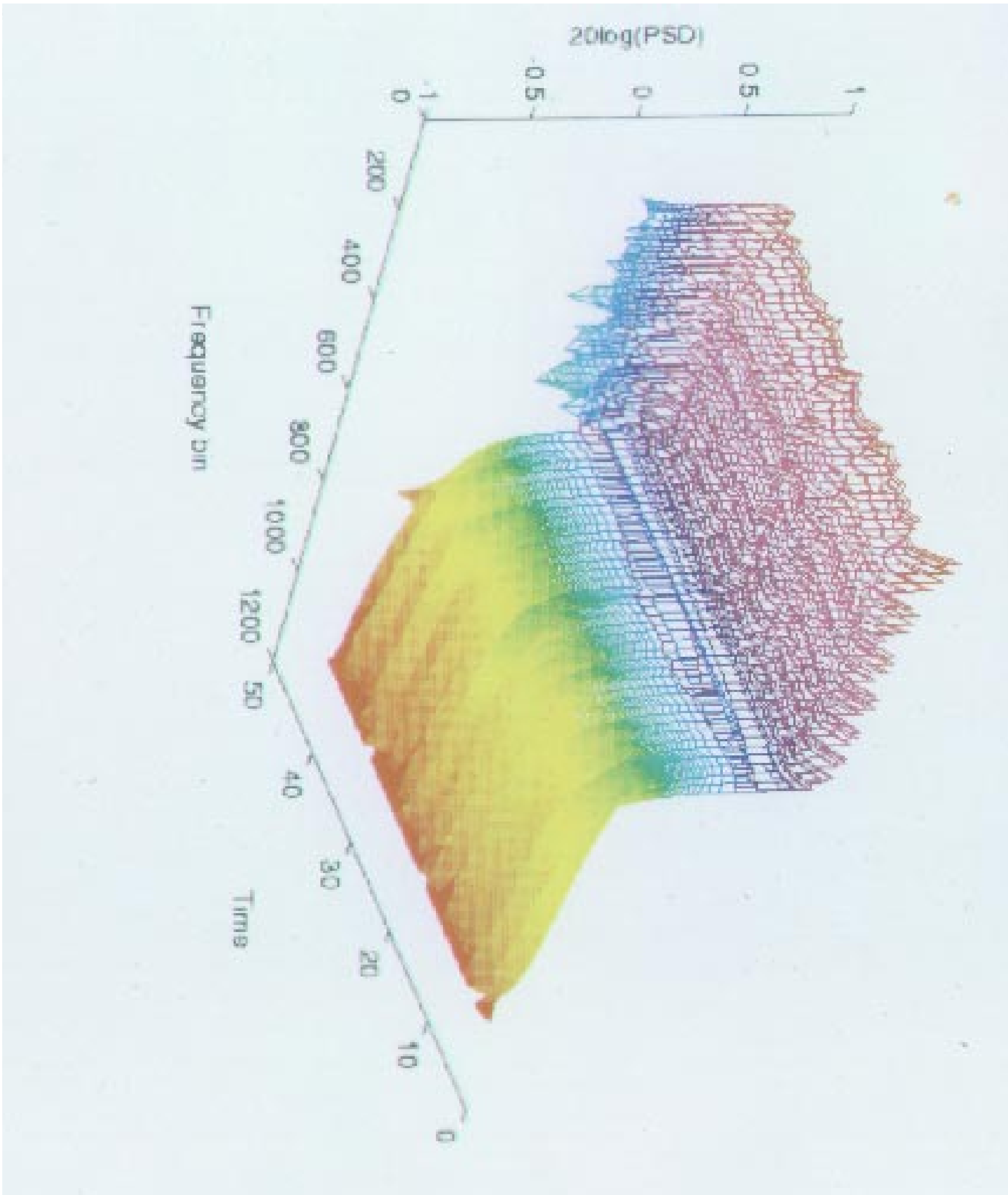
- Frequency spectrum of the waveguide model can be compared to that of a violin with no resonating body.
- The spectrum of the two signals should change in time similarly.
- Frequency spectrum must be analyzed over time rather than taking a single picture of the spectrum.

Subjective Measurements:

- It is not only important that the model be mathematically correct, but it must also have a pleasing sound.


Harmonic Decay
20log(PSD)
Frequency bin
Time


Harmonic Decay
20log(PSD)
1
0.5
0
-0.5
-1
0
Frequency bin
2000
4000
6000
8000
10000
25
20
15
10
5
0
Time


20log(PSD)
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-0.5
0
0.5
1
0
200
400
600
800
1000
1200
Frequency bin
50
40
30
20
10
0
Time

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WHAT HAVE WE SHOWN?

- Real time physical modeling is possible.
- Efficient models can be developed from the mathematical description of the physical system.
- The particular description can be tailored to specific instruments.
- Describing the physical characteristics of the instrument allows real time control over the characteristics of the sound.
- The digital waveguide can be broken down into functional blocks that described as standard elements of digital signal processing.

FUTURE RESEARCH

- Bowed strings
- Models to take more parameters into account
- Including the resonating cavity

REFERENCES

1. Bladier, Benjamin, On the Bridge of the Violoncello Compt. Rend., March 1960.
2. Campbell, Murray and Clive Greated, The Musician's Guide to Acoustics, Schirmer Books, New York, 1987.
3. Cremer, Lothar, The Influence of "Bow Pressure" on the Movement of a Bowed String, Catgut Acoustical Social Newsletter, No. 18, 13-19, 1972.
4. Cremer, Lothar, The Physics of the Violin, The MIT Press, Cambridge, Massachusetts, 1981.
5. Hutchings, C. M., The Physics of Violins, Scientific American, Inc., 1962.
6. Itokawa, Hideo, and Chihiro Kumagai, On the Study of Violin and Its Making, Report of the Institute of Industrial Science, 1952.
7. Morse, P.M., Vibration and Sound, American Institute of Physics for the Acoustical Society of America, Woodbury, New York, USA, 1976.
8. Ohlson, Harry F. Elements of Acoustical Engineering, D. Van Nostrand Co., Inc., New York, New York, 1947
9. Ousterhout, John K. Tcl and the Tk Toolkit, Addison-Wesley Publishing Company, Reading, Massachusetts, 1994.
10. Reid, Christopher E. and Thomas B. Passin, Signal Processing in C, John Wiley & Sons, Inc., New York, 1992.
11. Rorabaugh, C. Britton, Digital Filter Designer's Handbook: Featuring C Routines, Me Graw - Hill, Inc., Blue Ridge Summit, PA, 1993.
12. Smith III, Julius O., "Physical Modeling Using digital Waveguides," Computer Music Journal, vol. 16, No. 4, pp 74-91. Winter 1992.

Real-Time Speech Endpoint Detector

by

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ABSTRACT

- Accurate endpoint detection is a necessary capability for efficient construction of speech databases based on field recordings.
- This project implements a family of endpoint detection algorithms which uses signal features based on energy, zero-crossing rate.
- An objective evaluation paradigm has been developed to compare the endpoint detection algorithms.
- A reference speech database has been created as support for the evaluation methodology.
- Our implementation makes extensive use of object-oriented concepts and data-driven programming techniques.

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Motivation for Developing Endpointer

- ❏ To provide a real-time version of the ISIP energy based algorithm.
- ❏ To provide a comprehensive comparison of several popular algorithms.
- ❏ Accurate endpoint detection is a key component in the development of speech databases.

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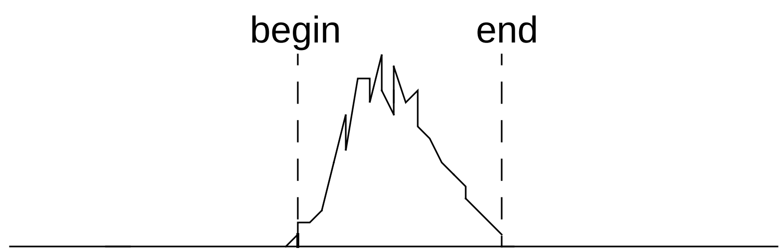
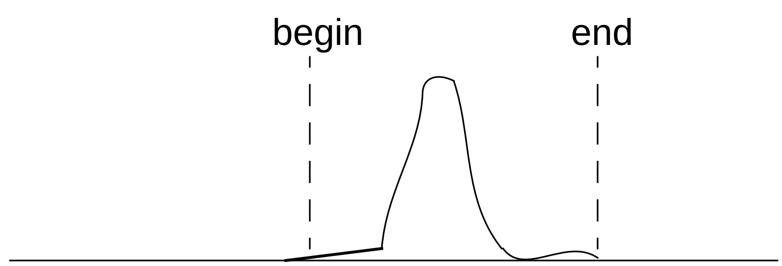
DIMENSIONS OF THE PROBLEM

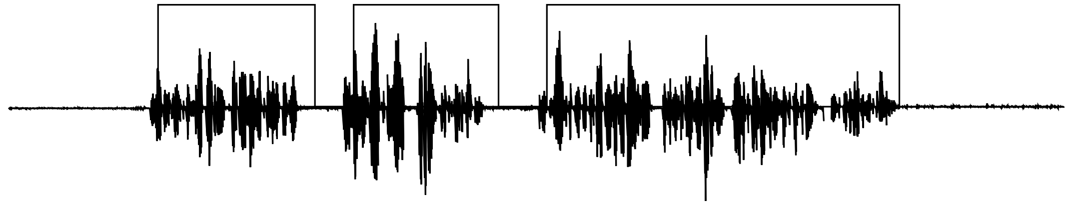
- ❑ Signal to Noise Ratio

- ❑ Energy

- ❑ Zero Crossings







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TODAY'S TECHNOLOGY

- Hidden Markov Models

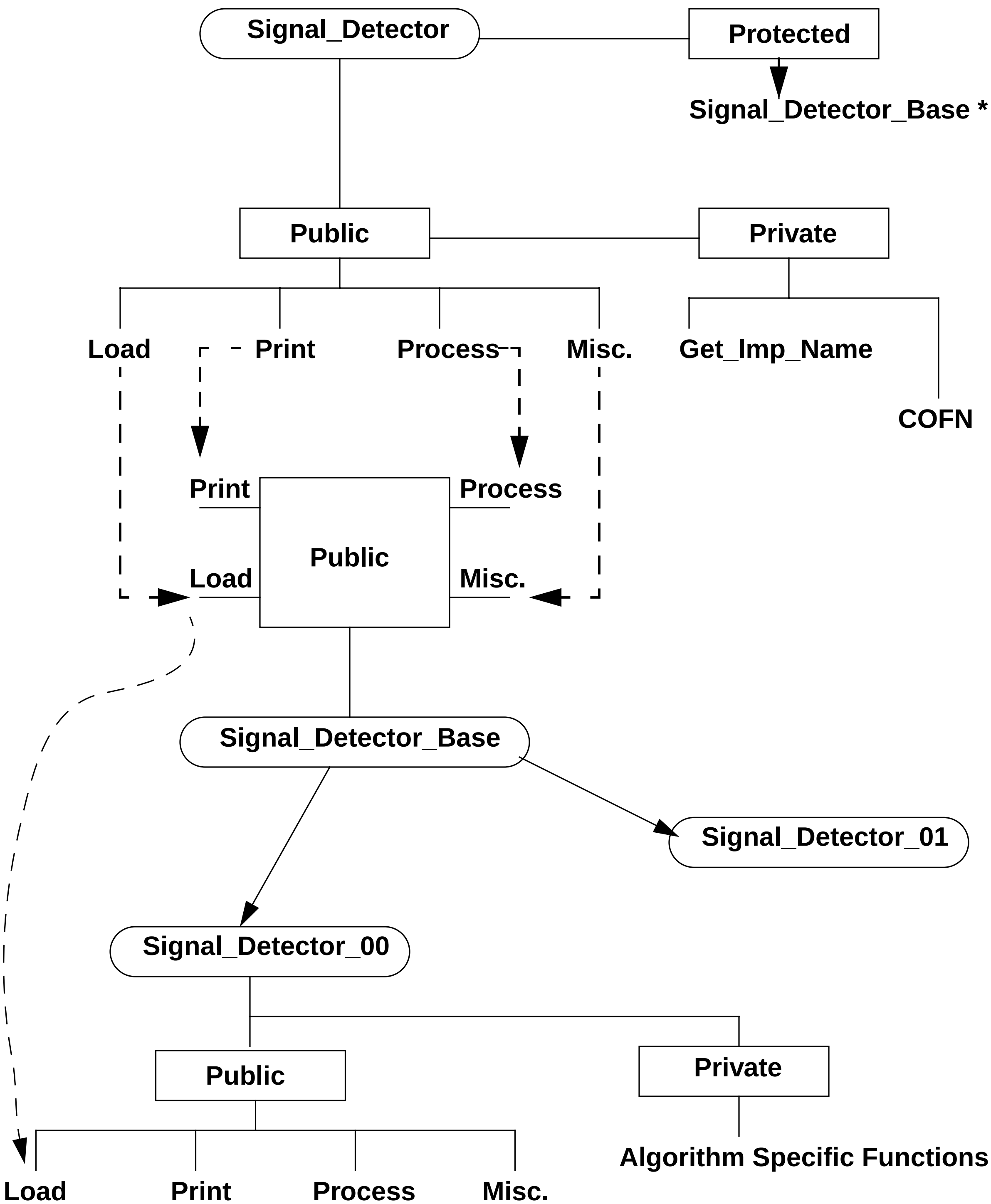
  A recognition strategy that makes use of a stochastic model of speech production. Each word in the recognition vocabulary is represented by a stochastic model.

- Neural Networks

  Solutions can add massively parallel computing strategies. Neural Networks can adapt and learn which is extremely useful in processing and recognizing speech. It tends to be more robust and fault tolerant.

- When using Hidden Markov Models and Neural Networks, endpoint detection is not used since automatic time alignment is done.

Class Structure



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Calculations

Energy

- preemphasis filter:

$$y(n) = x(n) - \alpha x(n-1) \qquad \alpha \rightarrow preemphasis - factor$$

- Energy Equation:

$$energy = (1/N)[\sum x^2(n) - \langle 1/N \rangle [\sum x(n)]^2]$$

Zero Crossings

- preemphasis filter:

$$y(n) = x(n) - \alpha x(n-1) \qquad \alpha \rightarrow preemphasis - factor$$

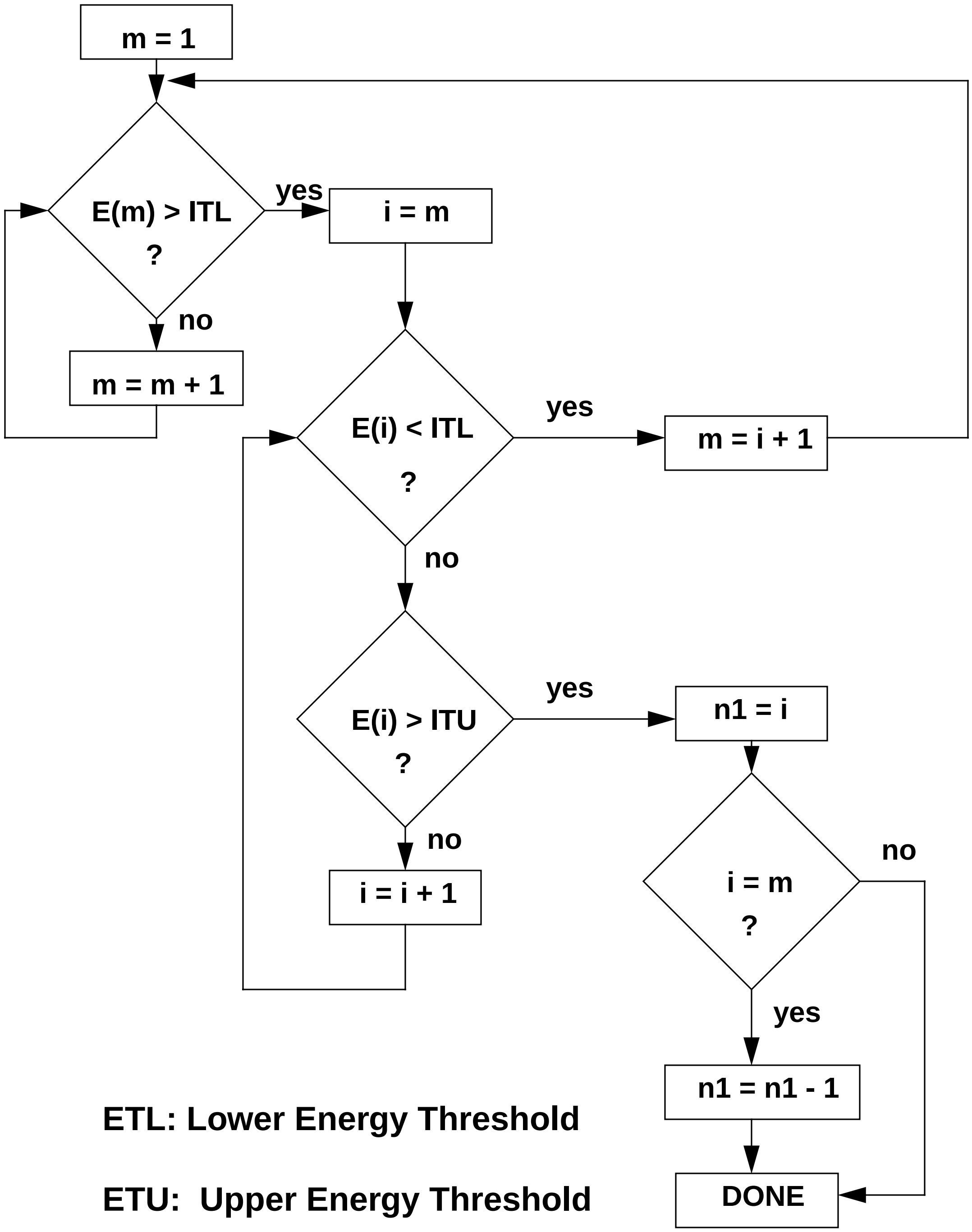
- Zero Crossing Rate

$$\sum_N [\mathrm{sgn}[x\langle n \rangle] - \mathrm{sgn}[x(m-1)]]w(n-m)$$

$$\mathrm{sgn} = 1 \qquad x(n) \geq pzcthresh$$
$$\mathrm{sgn} = -1 \qquad x(n) \leq nzcthresh$$

$$w(n) = 1/(2N) \qquad 0 \leq n \leq N-1$$
$$w(n) = 0 \qquad otherwise$$

Algorithms



ETL: Lower Energy Threshold

ETU: Upper Energy Threshold

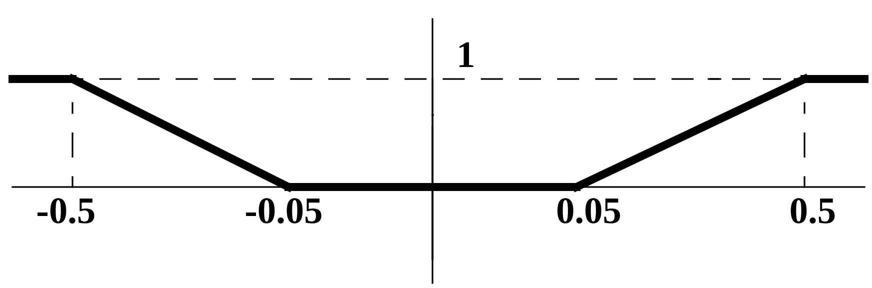
n1, n2: Endpoints

THE EVALUATION DATABASE

Table 1

File #	Data Type	Sample Data
1	single number	"zero" "two"
2	large number	"eighteen" "sixteen"
3	two word number	"65" "44"
4	very large number	"300,188"
5	phrase	"high definition television"
6	sentence pattern	"we hold these truths to be self-evident, that all men are created equal"
7	spontaneous speech	****************** (approx. 5 sentences)

THE EVALUATION METHODOLOGY



Scoring Function: +/- (2.22)error - 0.11

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Algorithm Results

Table 1:

Algorithm								
Speakers	sub	del	ins	Total	sub	del	ins	Total
Female01	.17	1	0	1.17	.16	1	0	1.16
Female02	.18	0	1	1.18	.19	0	1	1.19
Female03	.22	1	0	1.22	.21	1	0	1.21
Avg Fem	.19	2	1	1.19	.19	2	1	1.18
Male01	.32	1	0	1.32	.31	1	0	1.31
Male02	.33	0	1	1.33	.34	0	1	1.34
Male03	.18	0	0	0.18	.18	0	0	0.18
Avg Male	.28	1	1	0.94	.28	1	1	0.94
Utt Type								
Isolated Digits	0.11	0	0	0.11	0.12	0	0	0.12
Teens	0.10	2	1	3.10	0.06	2	1	3.06
Multi-Syl Dig	0.12	0	0	0.12	0.11	0	0	0.11
Long Dig	0.17	1	0	1.17	0.15	1	0	1.15
Short Sen	0.24	0	0	0.24	0.28	0	0	0.28
Long Sen	0.20	0	0	0.20	0.32	0	0	0.32
Spon Spc	0.73	0	0	0.73	0.71	0	0	0.71

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Algorithm Results

Algorithm				
Speakers	sub	del	ins	Total
Female01	.139	4	5	9.139
Female02	.174	4	1	5.174
Female03	.097	4	1	5.097
Avg Fem	.137	4	2.33	6.467
Male01	.224	8	3	11.22
Male02	.157	5	1	6.157
Male03	.213	9	2	11.21
Avg Male	.198	7.33	2	9.528

SUMMARY AND CONCLUSIONS

❏ Advantages of object oriented code:
- ☞ Extensibility
- ☞ Common framework

❏ Data with different SNRs are required to test the robustness of an algorithm. Our database has only data with medium SNR.

❏ Energy with zero-crossings performed better than energy alone.

❏ The Lowerre code
- ☞The code is only written to detect a single utterance
- ☞Hard coded parameters

❏ Endpoint detection is used for speech data trimming in the creation of databases.

❐ Because of automatic time alignment, endpoint detection is not always needed for speech recognition, but still plays an important role in database preparation.

REFERENCES

1. **L.R. Rabiner and M.R. Sambur, "An algorithm for determining the endpoints of isolated utterances," *Bell System Technical Journal*, vol. 54, pp. 297-315, 1975.**
2. **R. Tucker, "Voice Activity Detection Using a Periodicity Measure," *IEEE Proceedings, Part 1, Communications, Speech, Vision*, vol. 139, pp. 337-380, Aug. 1992.**
3. **J. Junqua, B. Mak, and B. Reaves, "A Robust Algorithm for Word Boundary Detection in Presence of Noise," *IEEE Transacitions on Speech and Audio Processing, vol. 2, No. 3, pp. 406-412, July 1994.***
4. **L.F. Lamel, L.R. Rabiner, A.E. Rosenberg, and J.G. Wilpon, "An Improved Endpoint Detector for Isolated Word Recognition," *IEEE Transactions on Acoustics, Speech, and Signal Processing, vol. ASSP-29, pp. 777-785, Aug. 1981.***
5. **B. Wheatley, et. al., "Robust Automatic Time Alignment of Orthographic Transcription with Unconstrained Speech," *IEEE Proceedings ICASSP,* vol. 1, pp. 533-536, 1992**
6. **D.P. Morgan and C.L. Scofield, *Neural Networks and Speech Processing*, Kluwer Academic Publishers, Norwell, Massachusetts, U.S.A., 1991.**

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ACOUSTIC NOISE CANCELLATION USING ADAPTIVE FILTERING TECHNIQUE

by

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ABSTRACT

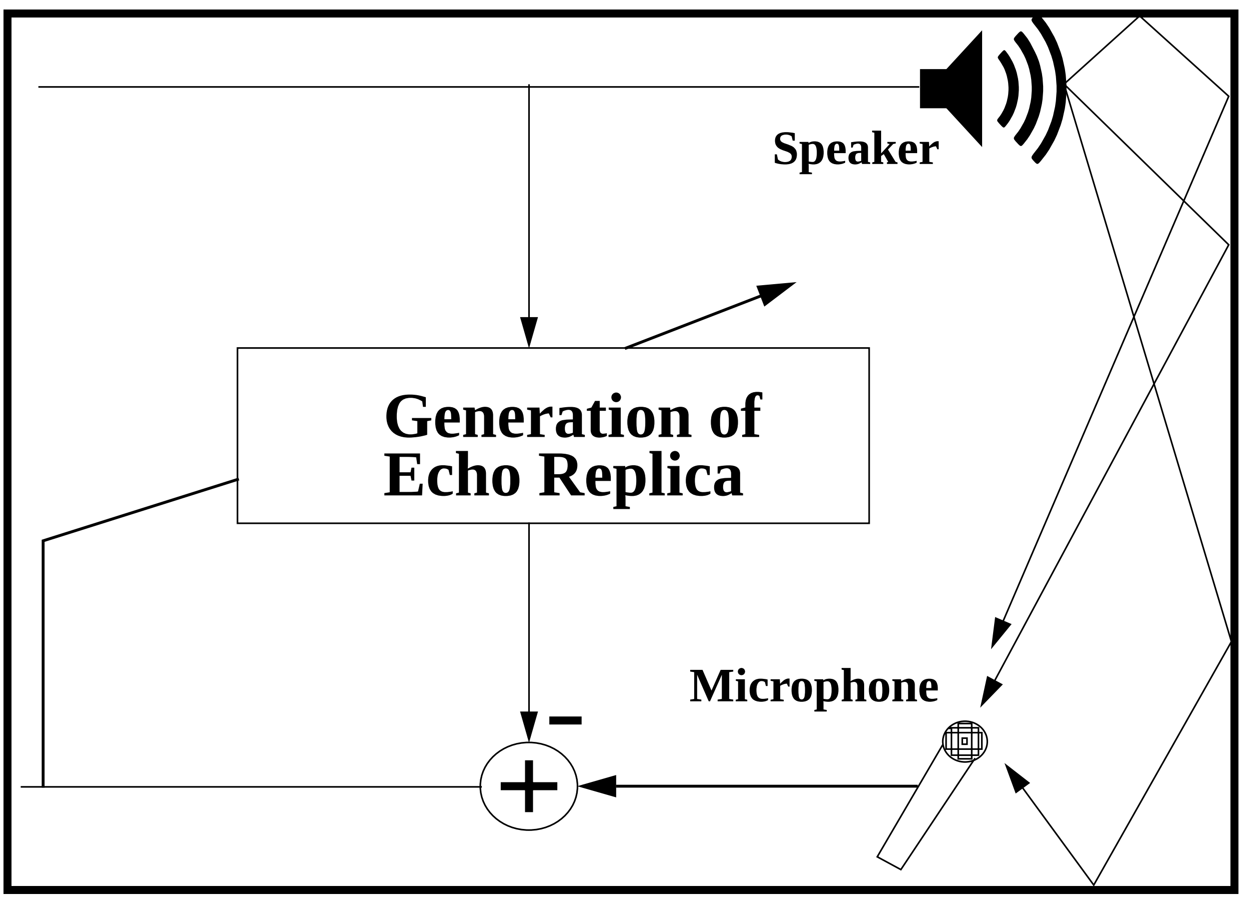
In an increasingly noisy society methods of reducing noise are becoming more important. Noise, or unwanted sound, may be reduced in an environment by two basic means: passive noise control and active noise control (ANC). Active noise control is a method of reducing noise by canceling a sound wave with an inverted copy of itself. This process works best in a simple environment: one in which the wavelength of the noise is long in relation to the dimensions of the space. ANC has been most successful in reducing noise in ducts and headphones (essentially one dimensional problems). This project will center around the application of an adaptive algorithm to implement echo cancellation (similar to existing technology used extensively in modern telephony).

ASPECTS OF THE PROBLEM

- **Low Frequency Reverberations are Bothersome**
  - ➨ **Impedes Conversation in Stairwell**
  - ➨ **Hinders Communication and Causes Fatigue**

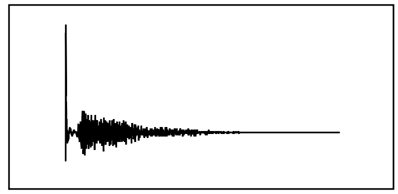
- **Two Possible Solutions:**
  - ➨ **Passive Noise Reduction--Not Much DSP**
  - ➨ **Active Noise Cancellation--Best Reduction in Noise**

WHAT IS THE THEORY BEHIND E.C.?



Steps in System Design

- **Estimate Echo Path Characteristic**



  - ⇒ **Select Adaptive Filter Type**
  - ⇒ **Select Algorithm for Adaptation**

- **Determine Length of Filter Necessary**

- **Determine the Variance of the Echo Path**
  - ⇒ **Does the Variance Require Continuous Updating of the Filter?**

- **Evaluate Performance of System on Artificial Data**

- **Apply System to Real Data**

CURRENT TECHNOLOGY

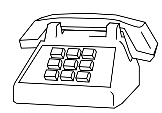
- **QuietQuiet^{TM} Acoustic Echo Cancellation by AT&T**

  - ⇒ **Echo Path Length:** **Up to 250 ms, nominally**
  - ⇒ **Adaptive Echo Cancellation** **>45 dB**
  - ⇒ **Convergence Rate** **30 dB/sec**
  - ⇒ **Full Duplex Operation Using Center Clipper Control**

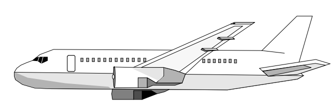
- **Acoustic Echo Control for TMS320C3x by PKI**

  - ⇒ **Echo Path Length:** **100 ms**
  - ⇒ **Adaptive Echo Cancellation** **45 dB**
  - ⇒ **Convergence Rate** **30 dB/sec**
  - ⇒ **Full Duplex Operation Using Center Clipper Control**

APPLICATIONS



- **Telephone Circuits/Full-Duplex Data Modem**



- **Aircraft Cabin Noise Attenuation**
- **Hearing Aid Feedback Reduction**
- **Determining Frequency Response of a Room** $\boxed{H(j\omega)}$

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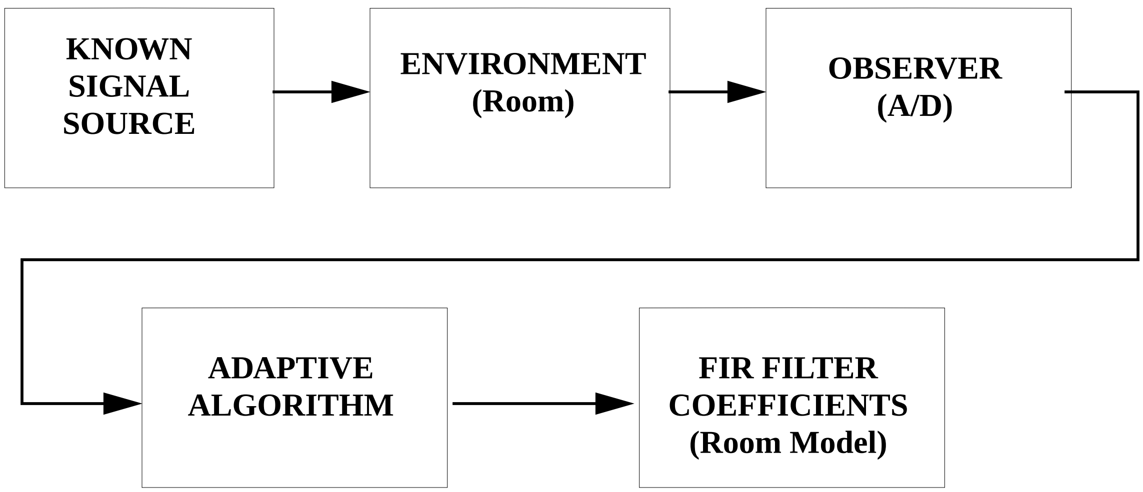
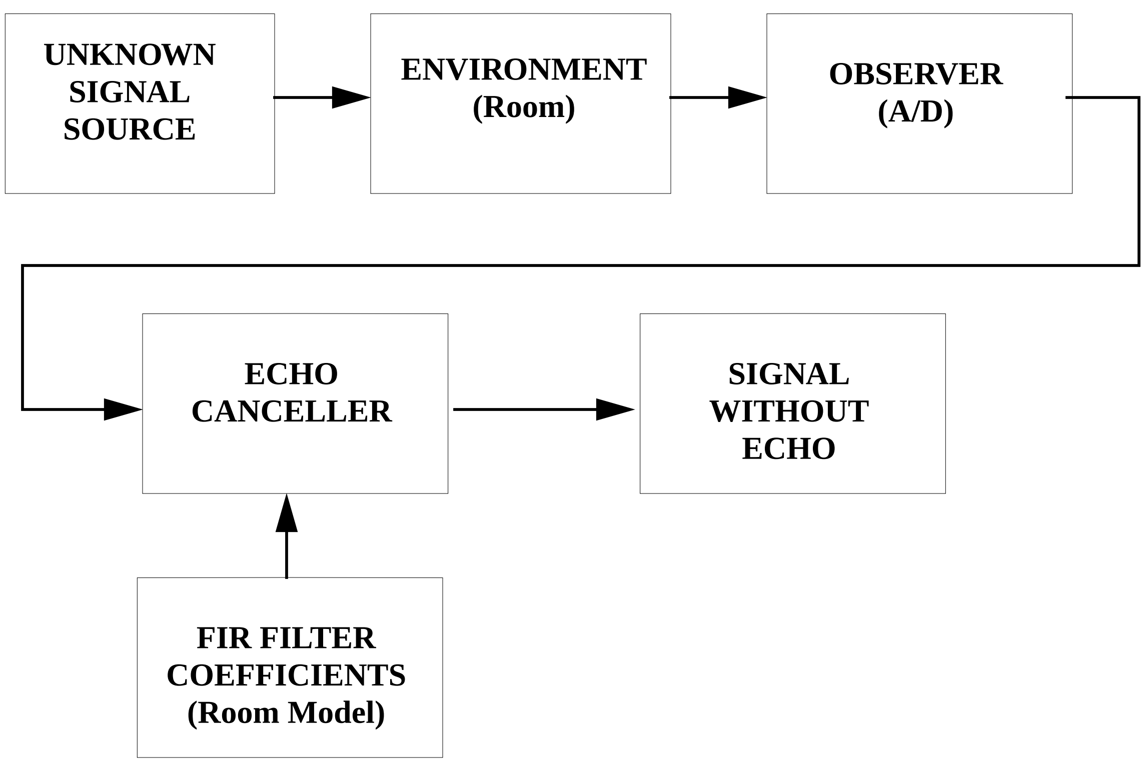
SYSTEM BLOCK DIAGRAM

Coefficient Adaptation



Acoustic Echo Canceler



MATHEMATICAL ANALYSIS

The reflected echo signal r(i) at time i is given by the convolution of the far-end reference signal y(i) and the discrete representation h_k of the impulse response of the echo path:

$$r(i) = \sum_{k=0}^{N-1} h_k y(i-k)$$

Since Linearity and a finite duration N of the echo-path response have been assumed, an echo canceller with N taps adapts the N coefficients a_k of its transversal filter to produce a replica of the echo r(i) defined as follows:

$$\hat{r}(i) = \sum_{k=0}^{N-1} a_k y(i-k)$$

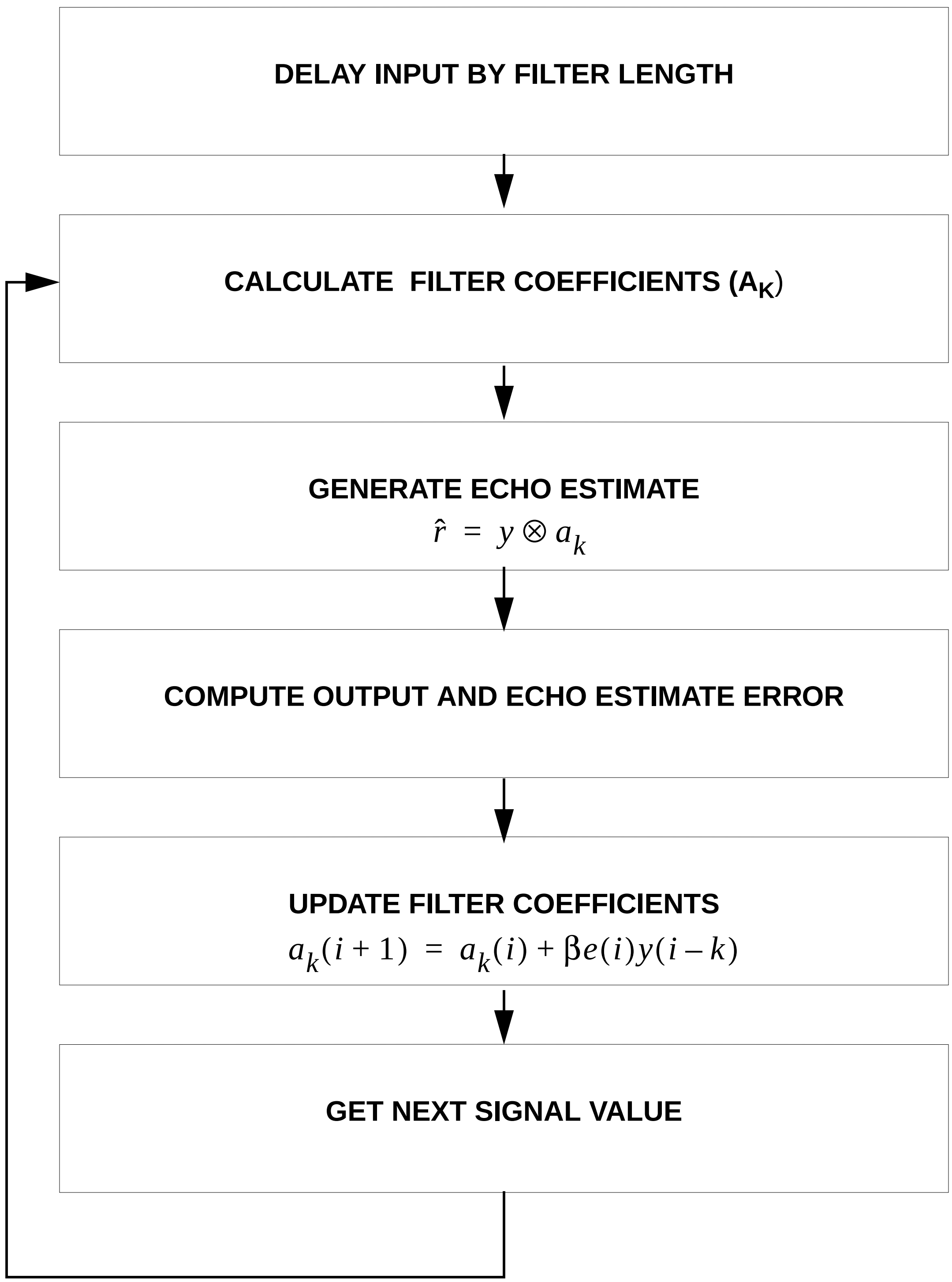
In General, the echo-path impulse response h_k is unknown and may vary slowly with time. Hence, a closed-loop coefficient adaptation algorithm is required to minimize the average or mean-squared error (MSE) between the echo and its replica. The update equation for the coefficients is given by:

$$a_k(i+1) = a_k(i) + 2\beta\frac{1}{M}\sum_{m=0}^{M-1} e(i-m)y(i-m-k)$$

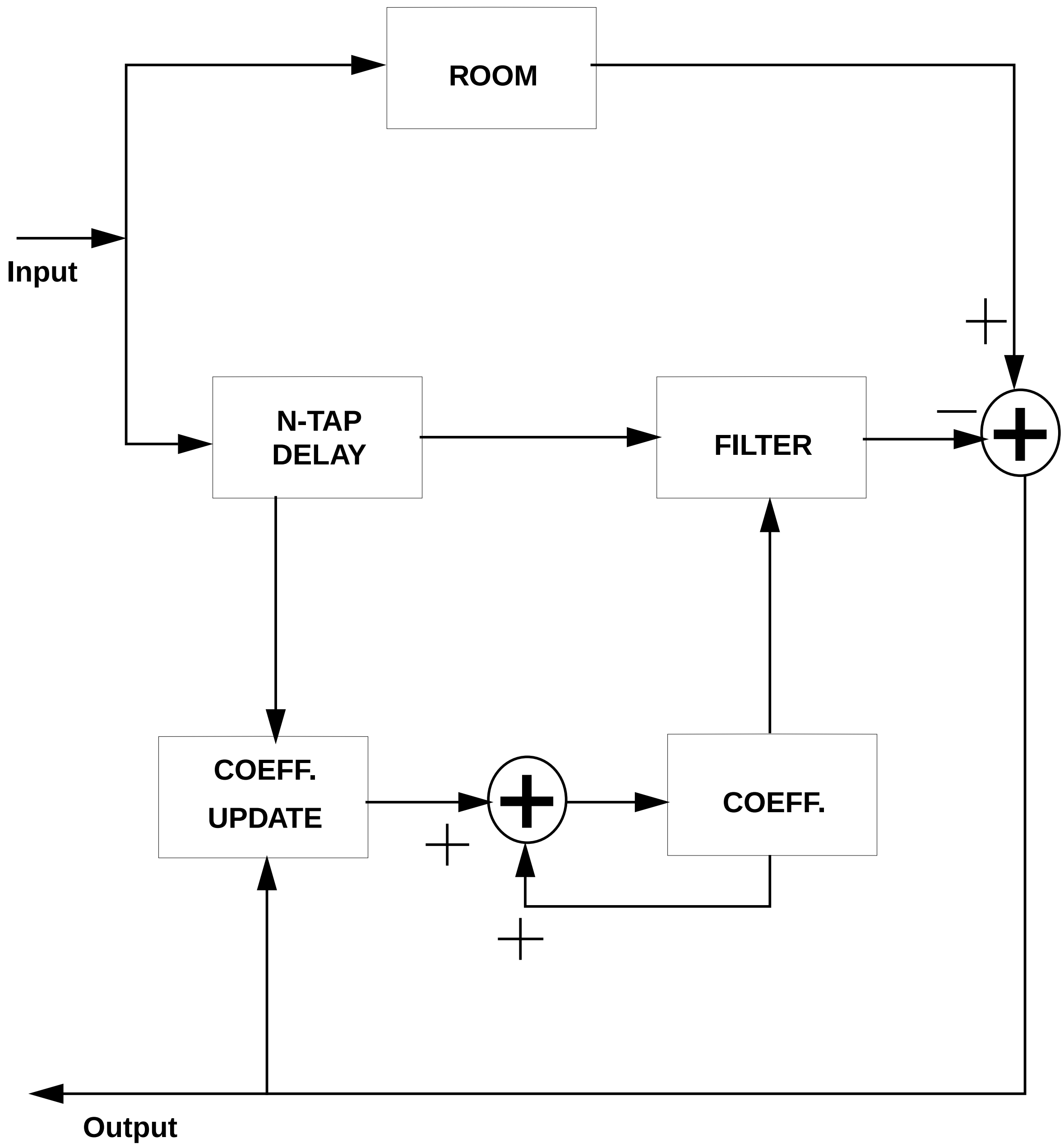
The closed loop gain of the system is normalized to the power in the input signal at the current time. In most common algorithms, the power is estimated using an IIR filter of the signal:

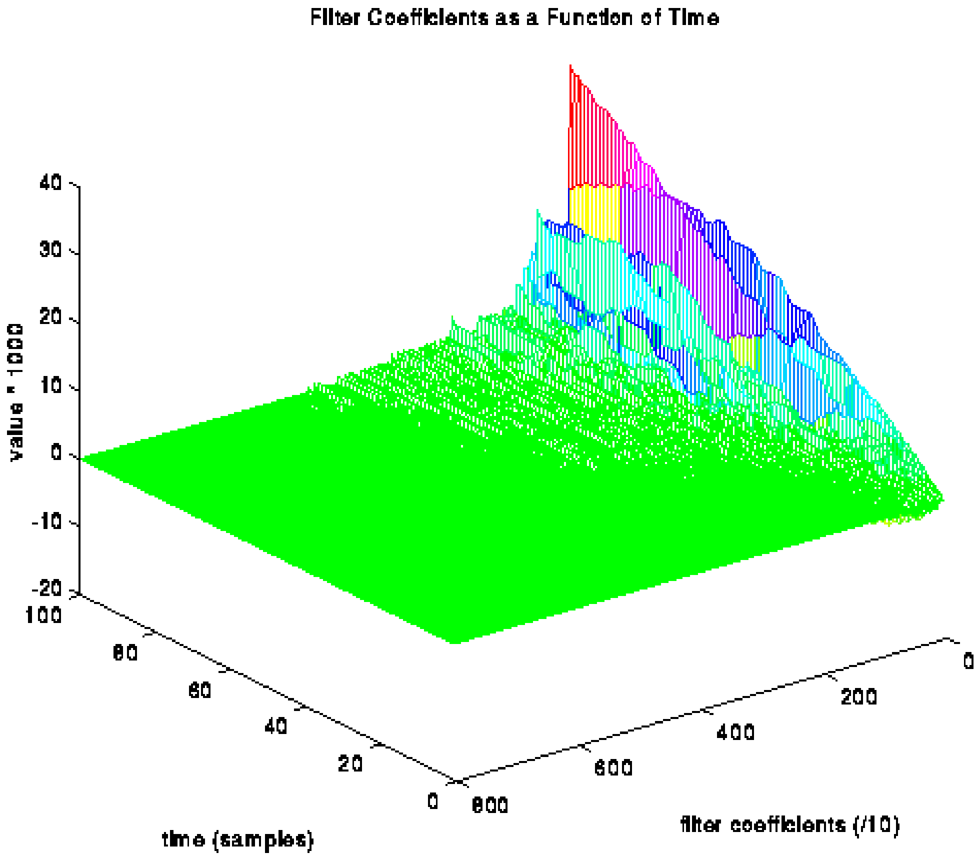
$$2\beta = 2\beta(i) = \frac{\beta_1}{P_y(i)}$$

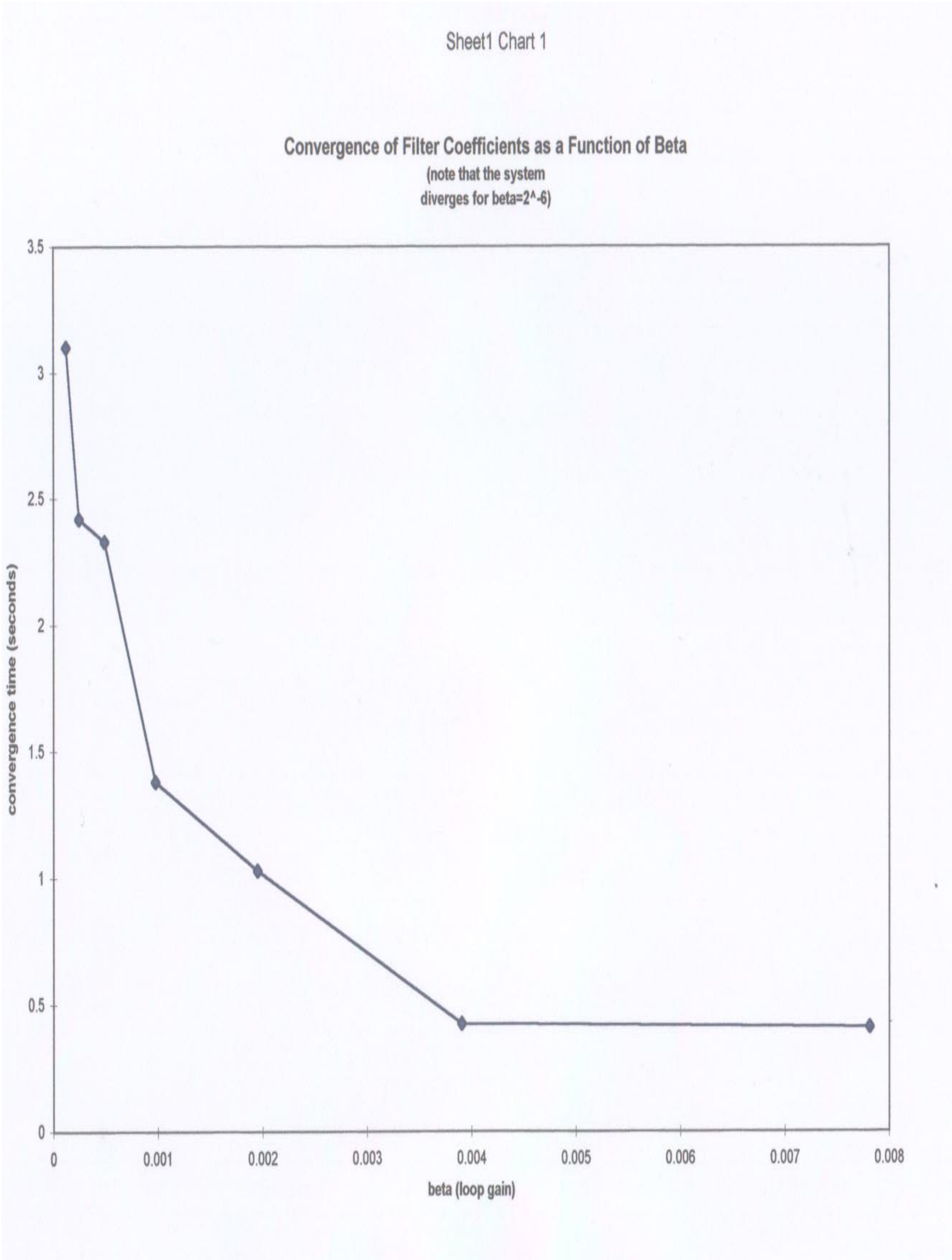
BASIC ADAPTIVE LMS ALGORITHM

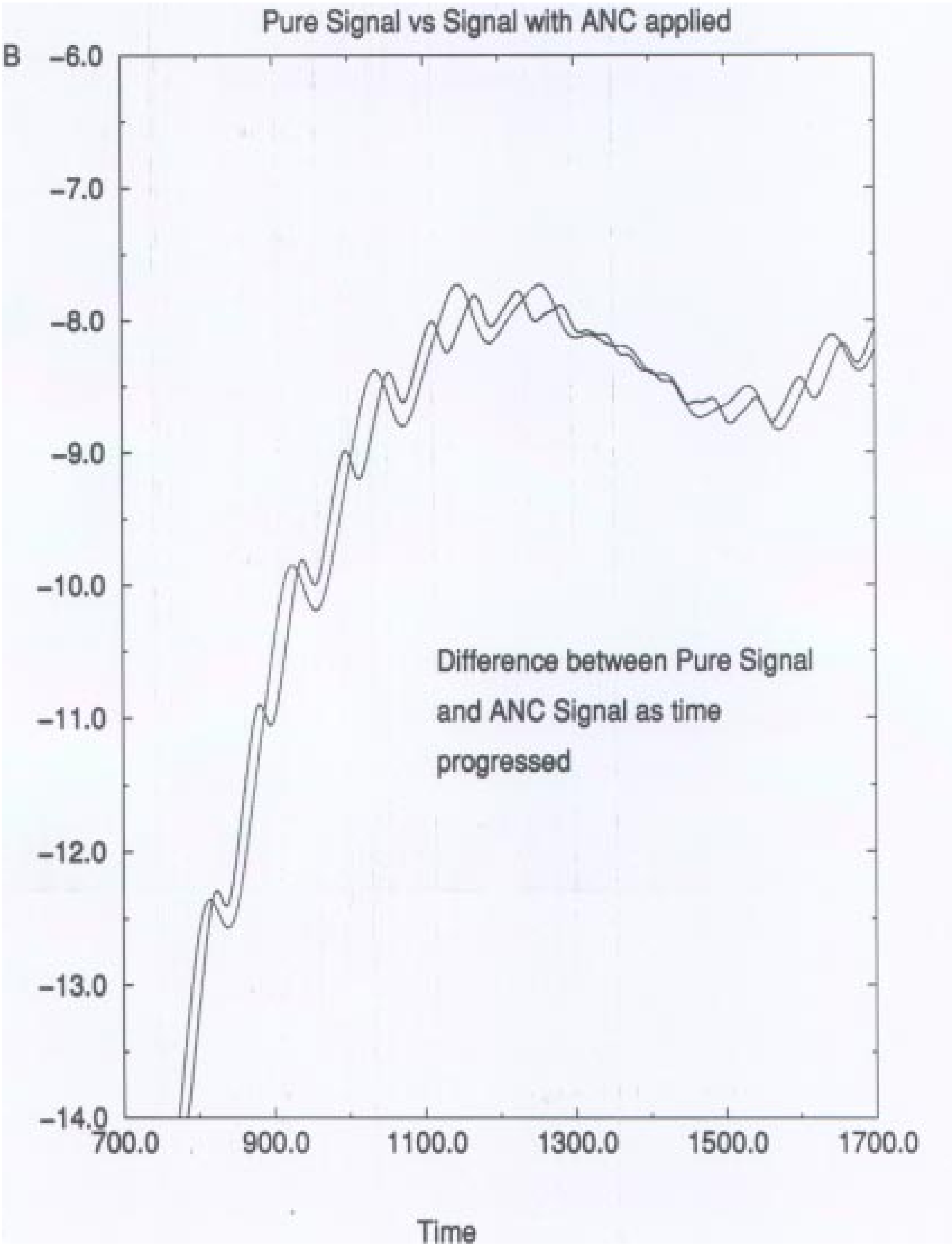


ECHO CANCELLATION BLOCK DIAGRAM




Filter Coefficients as a Function of Time
value * 1000
40
30
20
10
0
-10
-20
100
80
60
40
20
0
time (samples)
800
600
400
200
0
filter coefficients (/10)


Sheet1 Chart 1
Convergence of Filter Coefficients as a Function of Beta
(note that the system
diverges for beta=2^-6)
convergence time (seconds)
3.5
3
2.5
2
1.5
1
0.5
0
0
0.001
0.002
0.003
0.004
0.005
0.006
0.007
0.008
beta (loop gain)


Pure Signal vs Signal with ANC applied
B
-6.0
-7.0
-8.0
-9.0
-10.0
-11.0
-12.0
-13.0
-14.0
700.0
900.0
1100.0
1300.0
1500.0
1700.0
Time
Difference between Pure Signal and ANC Signal as time progressed

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EVALUATION OF SOFTWARE

- **COLLECTED DATA FROM ROOM**
  - ⇒ 2 CHIRPS (0-500 kHz, 0-20 kHz)
  - ⇒ 2 IMPULSES (100 Hz, 1kHz)
  - ⇒ 4 SINUSOIDS (100 Hz, 1kHz, 10kHz)
  - ⇒ VARIOUS SPEECH DATA
- **EXAMINED SPECTRUM OF ROOM RESPONSE**
- **TESTED ECHO CANCELLATION ALGORITHM**
  - ⇒ USING SIMPLE INPUTS AND IMPULSE RESPONSE
  - ⇒ USING PERIODIC IMPULSES
  - ⇒ BY APPLYING SYNCHRONIZED CHIRP AND ROOM RESPONSE
  - ⇒ BY CONVOLVING IMPULSE RESPONSE WITH SPEECH TO GET COMPLEX ECHO AND THEN APPLYING THE ECA
- **PERFORMED DFT'S ON ECHO SIGNAL AND IMPULSE RESPONSE**
- **APPLIED ECHO TO IIR FILTER WITH COEFF'S DETERMINED BY ECA**
- **PERFORMED DECONVOLUTION IN MATLAB TO FILTER ECHOS FROM SIGNAL**

Summary

- **The filter adaptation algorithm performed as expected.**
- **Studies on the algorithm indicate that on real data, loop gains of 2^{-7} are acceptable--yielding a convergence time of ~0.4 seconds.**
- **SNR ratios greater than 30 dB are well within reason.**
- **Coefficients need to be adapted only during the first second.**
- **'All' echoes can be cancelled if the signal generating the echoes is known exactly, and the echo path is suitably short.**
- **The echo canceller can take into account changing echo characteristics only if the signal is known exactly.**
- **Theoretically, once the coefficients are known for a particular echo characteristic, an analyzer (IIR) filter could be constructed that would filter out the echoes without knowing the signal, but any change in the system would require the filter coefficients to be updated.**
- **In a similar manner, all noise could be cancelled at a specific point using the same adaptive filter. Selective frequencies could be cancelled by banding the input data.**
- **The linked list data structure, the constant adaptation of the coefficients, the large tap size, floating point operations, and the generality of the program limited the speed of the echo canceller significantly. The canceller was one order of magnitude slower than real-time for 150 taps.**

RESULTS

- **The first major echo in the room was recorded at approximately 10 ms followed by a major reverberation at about 50 ms.**
- **Our figure of merit indicates a 6dB difference between the signal and the output of the canceller after only 0.4 seconds.**

REFERENCES

Echo Cancellation Algorithms:

1. Murano, Kazuo; Unagami, Shigeyuki; Amano, Fumio; “Echo Cancellation and Applications,” in *IEEE Communications Magazine, v. 28 pp. 49-55, January ‘90.*
2. Petillon, Thierry; Gilloire, Andre; Theodoridis, Sergios; “The Fast Newton Transversal Filter: An Efficient Scheme for Acoustic Echo Cancellation in Mobile Radio,” in *IEEE Transactions on Signal Processing*, v. 42 pp. 2134-44 May ‘94.
3. Gilloire, Andre; Vetterli, Martin; “Adaptive Filtering in Subbands with Critical Sampling: Analysis, Experiments, and Application to Acoustic Echo Cancellation,” *IEEE Transactions on Signal Processing*, v. 40 pp. 1862-75, August ‘92.
4. van de Kerkhof, Leon M.; Kitzen, Wil J. W.; “Tracking of a Time-Varying Acoustic Impulse Response by an Adaptive Filter,” in *IEEE Transactions on Signal Processing*, v. 40 pp. 1285-94, June ‘92.

Echo Cancellation Applications:

5. Clark, Robert; “Active Damping on Enclosed Sound Fields Through Direct Rate Feedback Control,” in *Acoustical Society of America*, pp. 1710-16, March 1995.
6. Gingell, M.J.; Hay, B.G.; Humphrey, L.D. “A Block Mode Update Echo Canceller Using Custom LSI,” *GLOBECOM Conference Record*, v. 3, 1394-97 (November 1983).
7. Kates, James M.; “Feedback Cancellation in Hearing Aids: Results from a Computer Simulation,” in *IEEE Transactions on Signal Processing*, v 39 pp. 553-62, March ‘91.
8. Kuo, Sen M.; Pan Zhibing; “Distributed Acoustic Echo Cancellation System with Double-Talk Detector,” in *The Journal of the Acoustical Society of America*, v. 94 pp. 3057-60 December ‘93
9. Messerschmitt, David; Hedberg, David; Cole, Christopher; Haoui, Amine; Winship, Peter; “Digital Voice Echo Canceller with a TMS32020,” in *Digital Signal Processing Applications with the TMS320 Family*, pp. 415-437, Texas Instruments, Inc., 1986.
10. Messerschmitt, David; “Echo Cancellation in Speech and Data Transmission,” in *IEEE Journal on Selected Topics in Communications*, SAC-2, No. 2, 283-303 (March 1984).
11. Picone, J.; Johnson, M.A.; Hartwell, W.T.; “Enhancing Speech Recognition Performance with Echo Cancellation,” in *Proceedings IEEE International Conference on Acoustics, Speech, and Signal Processing*, pp. 529-532, New York, New York, USA, April 1988.
12. Ruckman, C.E;, “Active Noise Control FAQ,” Available via anonymous ftp as from ftp://rtfm.mit.edu/pub/usenet/news.answers/active_noise_control_faq.
13. Shieh, Chifong; Bai,Mingsian R.; “Active Noise Cancellation by Using The Linear Quadratic Gaussian Independent Modal Space Control,” *Acoustical Society of America.*, pp. 2664-2671, May 1995.

14 Sondhi, Man Mohan; Berkley, David A.; "Silencing Echoes on the Telephone .Network," in *Proceedings of the IEEE,* v. 68 pp. 948-63, August 1980.

15. Tokhi, M.O.; Leitch, R.R.; *Active Noise Control*, Oxford Science Publications, Clarendon Press 1992.

Digital Signal Processing:

16. Proakis, J.G.;. Manolakis, D.G.; *Digital Signal Processing: Principles, Algorithms, and Applications*, *2nd Edition*, Macmillan, New York, New York, USA, 1992.

Object-Oriented Software:

17. Schildt, Herbert; *Teach Yourself C, 2nd ed.*, McGraw Hill, Berkeley, California, USA, 1994.

Math Tools:

18. *The Student Edition of MATLAB,* Prentice Hall, Englewood Cliffs, New Jersey, USA, 1992.

On the Net/Current DSP Hardware:

19. http//www.ti.com/sc/docs/dsps/softcoop/ Software Overviews for several different echo cancellation algorithms.