Advanced Functions of Java-DSP

for use in Electrical and Computer Engineering Senior Level Courses

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Topics

This presentation uses Java-DSP to cover 3 areas of DSP relevant to the implementation of smartphones.

- Audio Content Analysis
- Echo Cancellation
- Transmit Amplifier Linearization

SenSIP

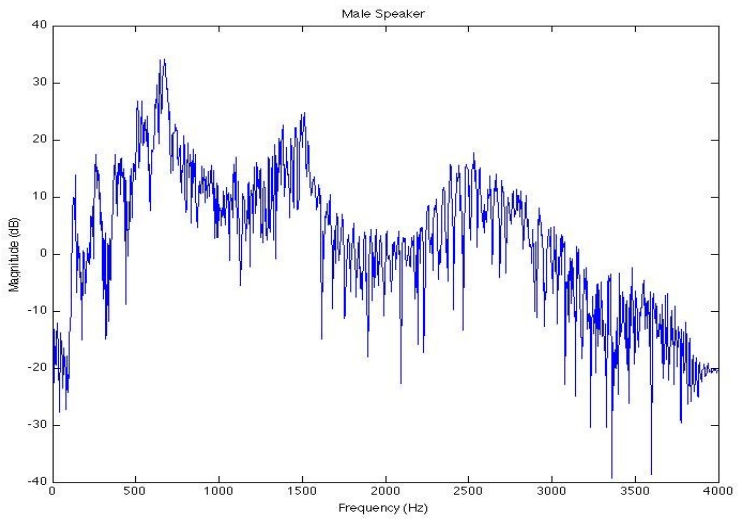
Audio Content Analysis

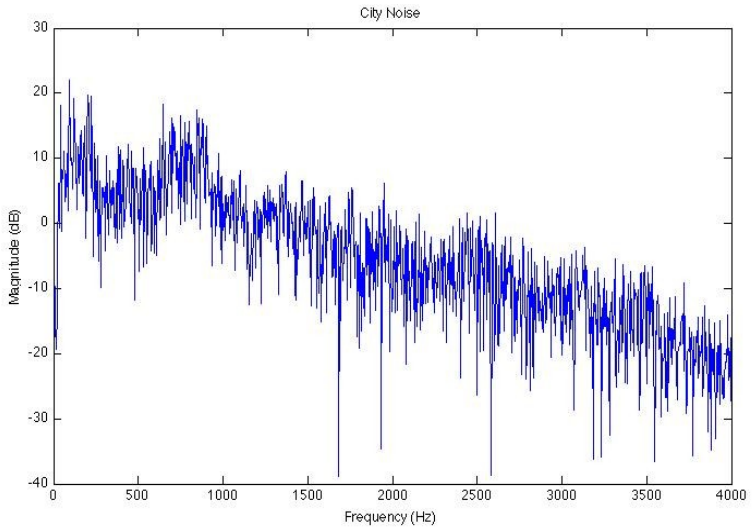
Speech and Music Applications

- Speech Recognition
- Speaker Recognition
- Music Genre Classification (Pandora, Last.FM)
- Audio Fingerprinting (Shazam application)
- Query-by-Humming (Melodis SoundHound)
- Environmental Soundscapes (Soundwalks)

Feature Extraction

- Features - We exploit the structure present in the temporal/spectral domain of audio signals to extract certain key characteristics that help us distinguish these signals amongst each other.





- By observation of the spectrum, distance between the peaks (pitch) and the position of the strongest peak (fundamental frequency) can be used as preliminary features for characterizing two different sounds.

Feature Extraction (Contd.)

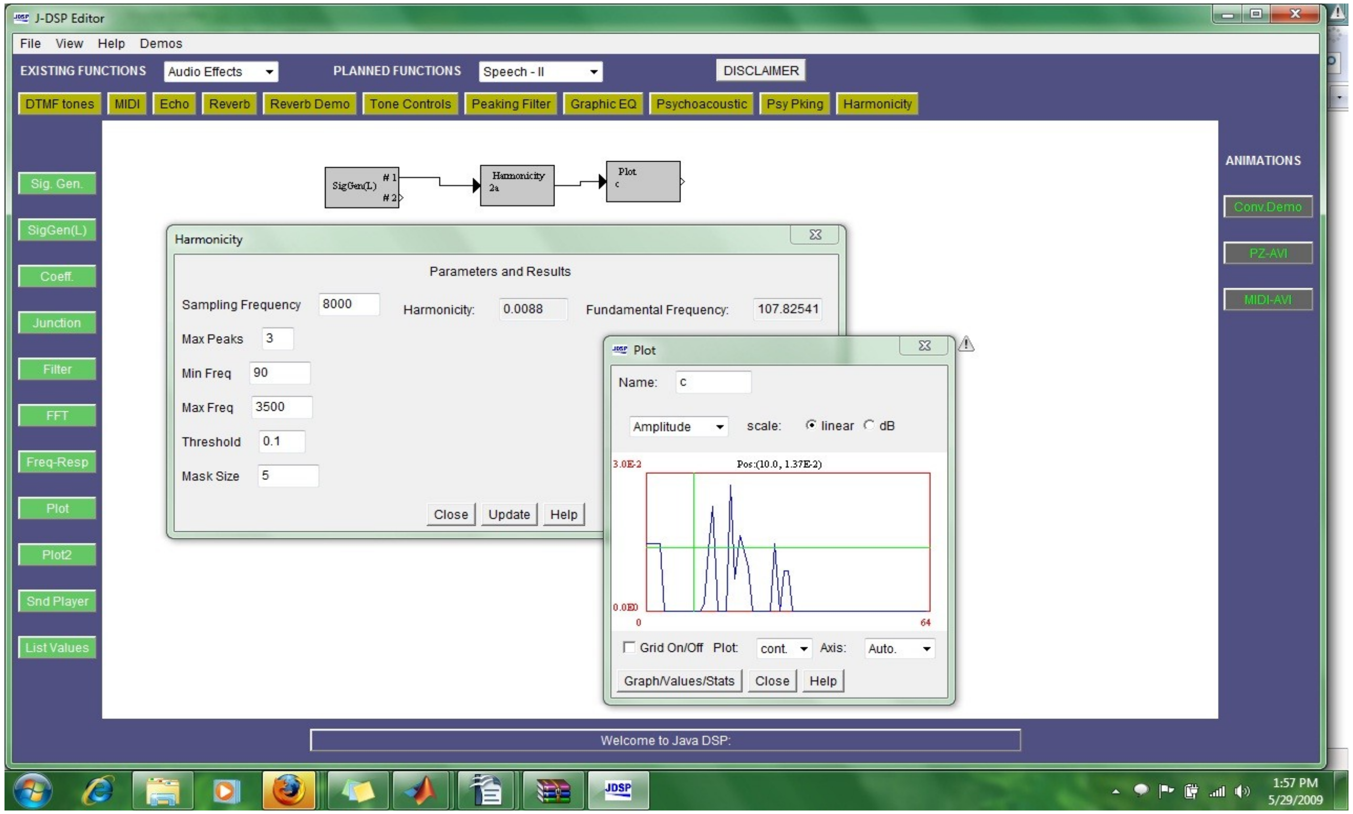
•Students can be taught to recognize such patterns in audio signals based on certain mathematical and signal processing techniques.

•Some important Features

Loudness:	Screaming into the microphone has higher loudness
Spectral Centroid:	Bright/shrill sounds have higher spectral centroid
Tonality:	Clean speech has higher tonality compared to noisy speech
Harmonicity:	Speech & music exhibit a harmonic structure in their spectrum

•Having obtained a basic set of features, we then build a ***supervised*** (SVM) or ***unsupervised*** (Clustering, Gaussian Mixture Models) learning system for classification and retrieval of these signals.

Implementation in J-DSP



EXISTING FUNCTIONS: Audio Effects
PLANNED FUNCTIONS: Speech - II

Harmonicity

Parameters and Results

Sampling Frequency: 8000
Max Peaks: 3
Min Freq: 90
Max Freq: 3500
Threshold: 0.1
Mask Size: 5
Harmonicity: 0.0088
Fundamental Frequency: 107.82541

Plot

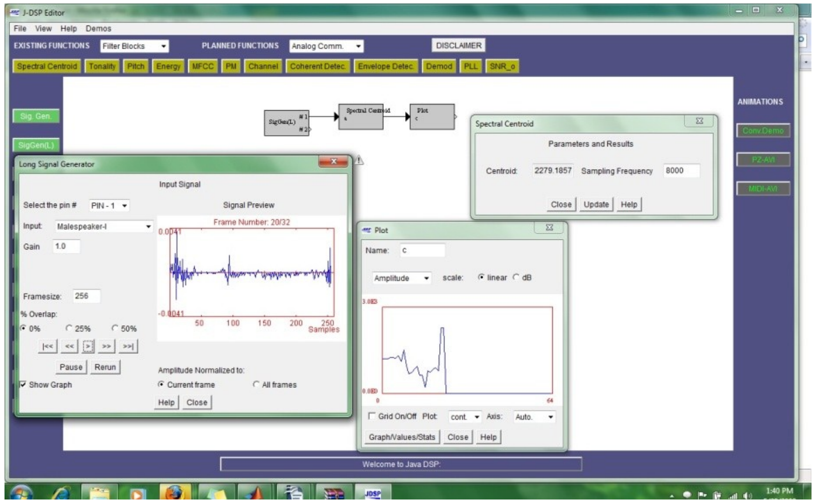
Name: c
Amplitude
scale: [✓ linear] dB
Grid On/Off
Plot: cont.
Axis: Auto.

Welcome to Java DSP:

Implementation of an algorithm to extract *harmonicity* in J-DSP. Equipped with a dialog box to tweak certain parameters and observe the results in a real-time plot, this toolbox helps students gain insight into the understanding of harmonicity in different kinds of audio signals.

Tutorials and Exercises

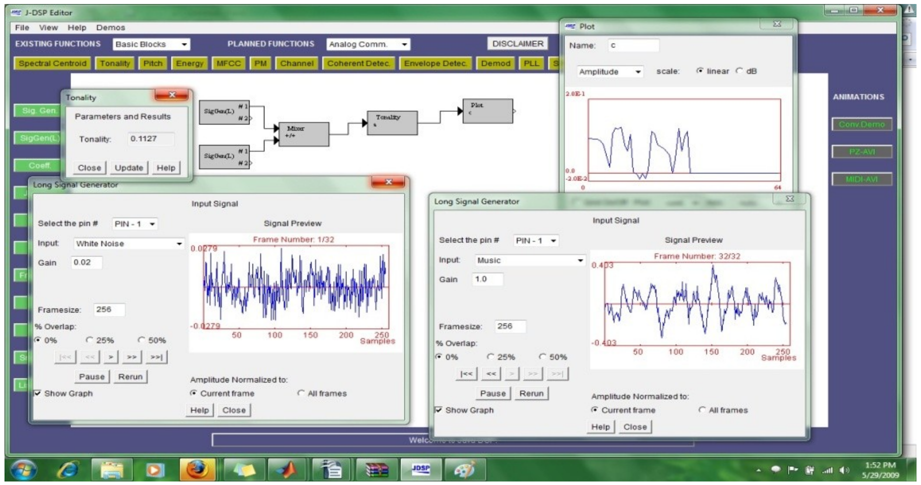
- A set of tutorials have been developed in J-DSP to give students a basic knowledge of pattern recognition and feature extraction.
- Exercises to extract *temporal energy, spectral centroid, pitch and tonality* have been developed.



Spectral Centroid — Parameters and Results: Centroid: 2279.1857; Sampling Frequency: 8000

Long Signal Generator — Select the pin #: PIN - 1; Input: Malespeaker-I; Gain: 1.0; Framesize: 256; % Overlap: 0%; Frame Number: 20/32; Amplitude Normalized to: Current frame; Show Graph

Plot — Name: c; Amplitude; scale: linear; Plot: cont.; Axis: Auto.

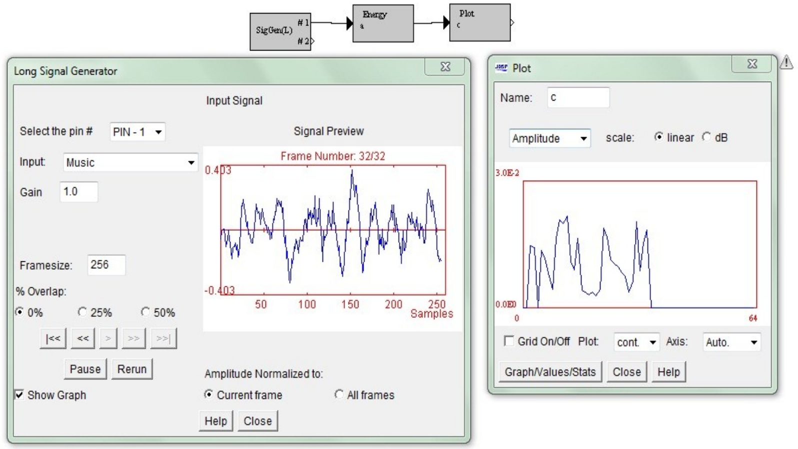


Tonality — Parameters and Results: Tonality: 0.1127

Long Signal Generator — Select the pin #: PIN - 1; Input: White Noise; Gain: 0.02; Framesize: 256; % Overlap: 0%; Frame Number: 1/32; Amplitude Normalized to: Current frame; Show Graph

Long Signal Generator — Select the pin #: PIN - 1; Input: Music; Gain: 1.0; Framesize: 256; % Overlap: 0%; Frame Number: 32/32; Amplitude Normalized to: Current frame; Show Graph

Plot — Name: c; Amplitude; scale: linear



Long Signal Generator — Select the pin #: PIN - 1; Input: Music; Gain: 1.0; Framesize: 256; % Overlap: 0%; Frame Number: 32/32; Amplitude Normalized to: Current frame; Show Graph

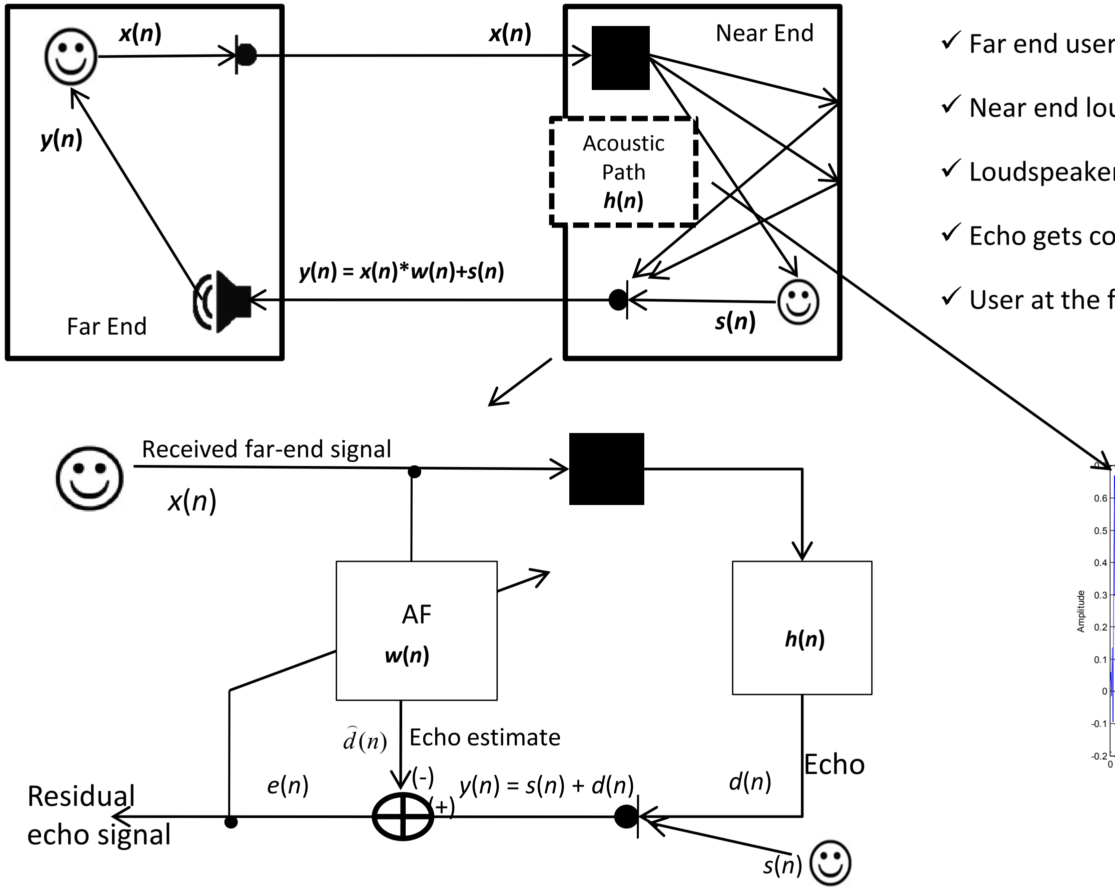
Plot — Name: c; Amplitude; scale: linear; Grid On/Off; Plot: cont.; Axis: Auto.

Topics

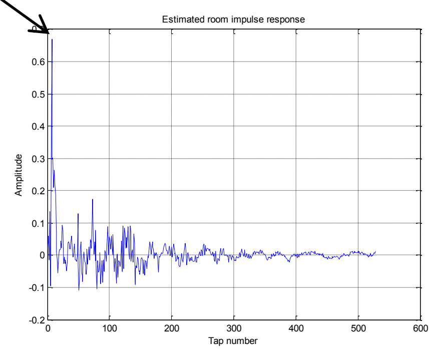
- Audio Content Analysis
- Acoustic Echo Cancellation
- Transmit Amplifier Linearization

Acoustic Echo Cancellation

The Acoustic Echo Problem

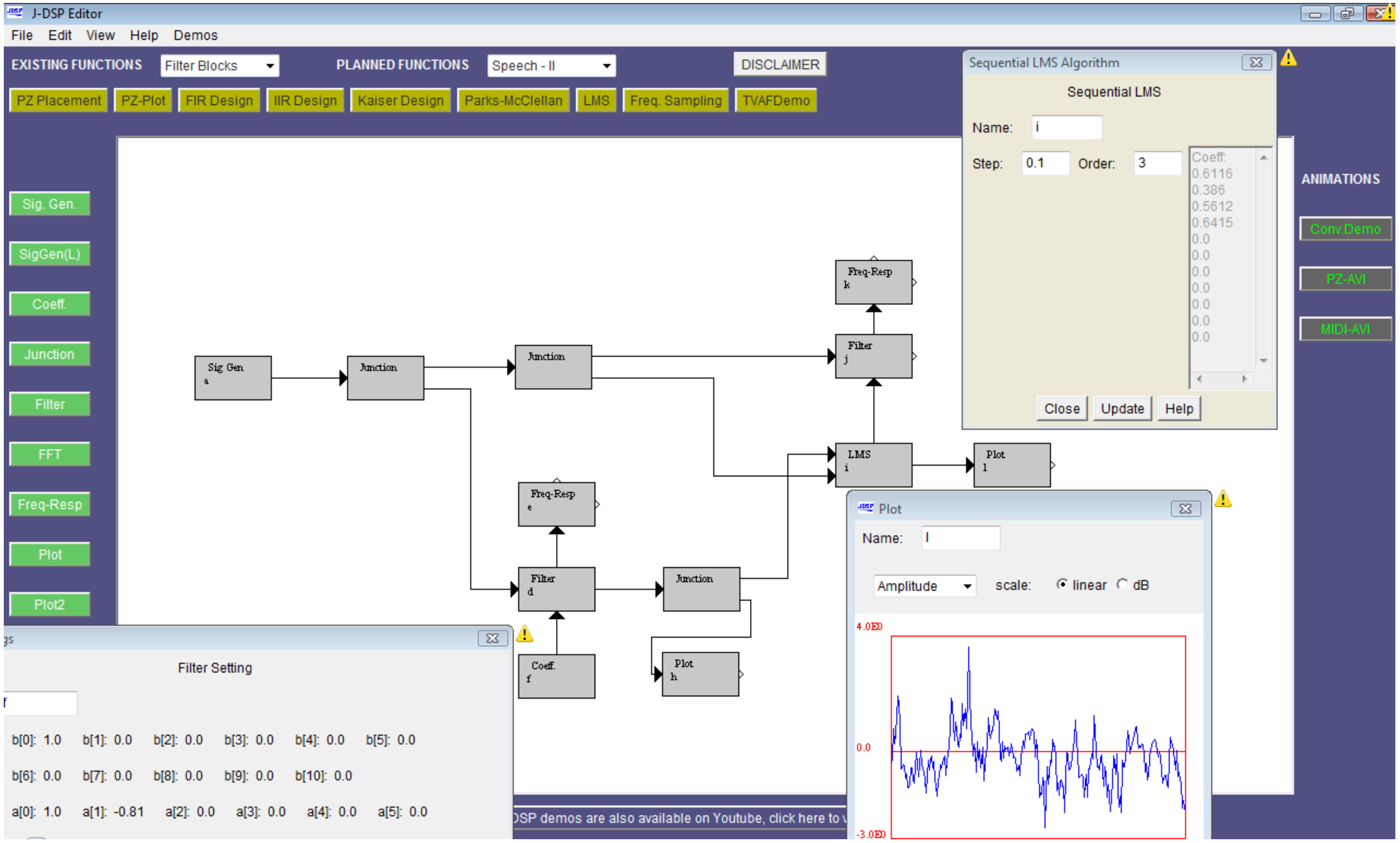


- ✓ Far end user speaks
- ✓ Near end loudspeaker plays
- ✓ Loudspeaker signal bounces-off the walls at the near-end
- ✓ Echo gets collected by the microphone at the near-end
- ✓ User at the far-end hears his/her own voice



The adaptive filter produces an estimate of acoustic room impulse response and calculates the echo contribution, which is then finally subtracted from the microphone signal.

JDSP Demonstration: Acoustic Echo Cancellation



Sequential LMS Algorithm

Sequential LMS

Name: i

Step: 0.1 Order: 3

```
Coeff:
0.6116
0.386
0.5612
0.6415
0.0
0.0
0.0
0.0
0.0
0.0
0.0
```

Close Update Help

Plot

Name: l

Amplitude scale: [✓ linear] dB

Filter Setting

f

```
b[0]: 1.0   b[1]: 0.0   b[2]: 0.0   b[3]: 0.0   b[4]: 0.0   b[5]: 0.0
b[6]: 0.0   b[7]: 0.0   b[8]: 0.0   b[9]: 0.0   b[10]: 0.0
a[0]: 1.0   a[1]: -0.81   a[2]: 0.0   a[3]: 0.0   a[4]: 0.0   a[5]: 0.0
```

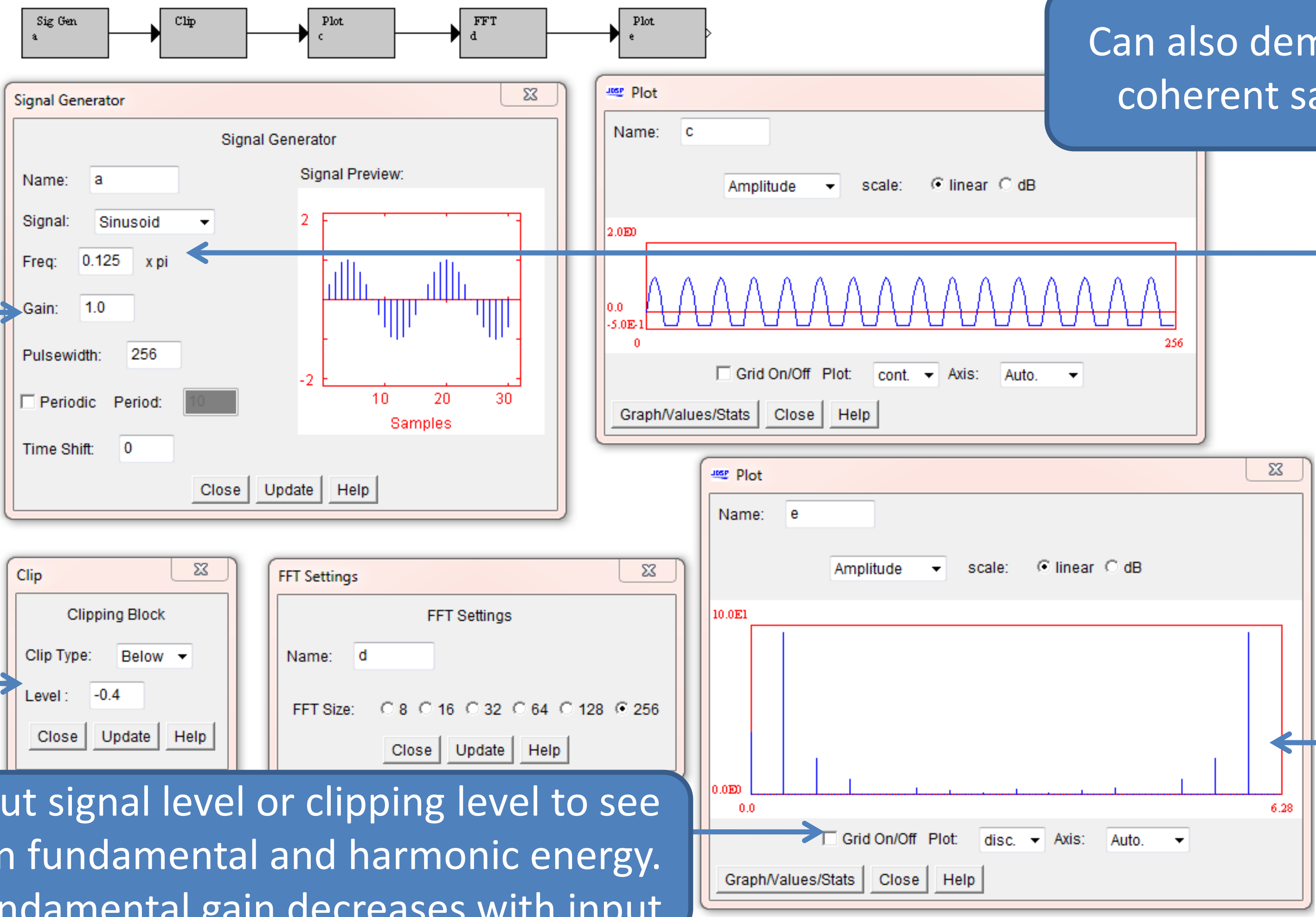
Topics

- Audio Content Analysis
- Echo Cancellation
- Transmit Amplifier Linearization

Mobile Radio Transmitters

- Two Conflicting Design Requirements
  - Power Efficient Amplifiers
    - Increase battery lifetime for mobile handsets
    - Decrease cooling requirements for base stations.
    - Usually operate in/near compression region
  - Highly linear amplifiers
    - Reduce distortion-caused spillover to adjacent channels
    - Allows Packing Channels Closely Together
    - Minimize transmitted signal error
- Non-linearity (clipping) introduces harmonics

JDSP Simulation:
How does Clipping Generate Harmonics?



Signal Generator

Signal Generator

Name: a

Signal: Sinusoid

Freq: 0.125 x pi

Gain: 1.0

Pulsewidth: 256

Periodic [ ] Period: 10

Time Shift: 0

Close Update Help

Plot

Name: c

Amplitude scale: [✓ linear] dB

Grid On/Off [ ] Plot: cont. Axis: Auto.

Graph/Values/Stats Close Help

Plot

Name: e

Amplitude scale: [✓ linear] dB

Grid On/Off [ ] Plot: disc. Axis: Auto.

Graph/Values/Stats Close Help

Clip

Clipping Block

Clip Type: Below

Level : -0.4

Close Update Help

FFT Settings

FFT Settings

Name: d

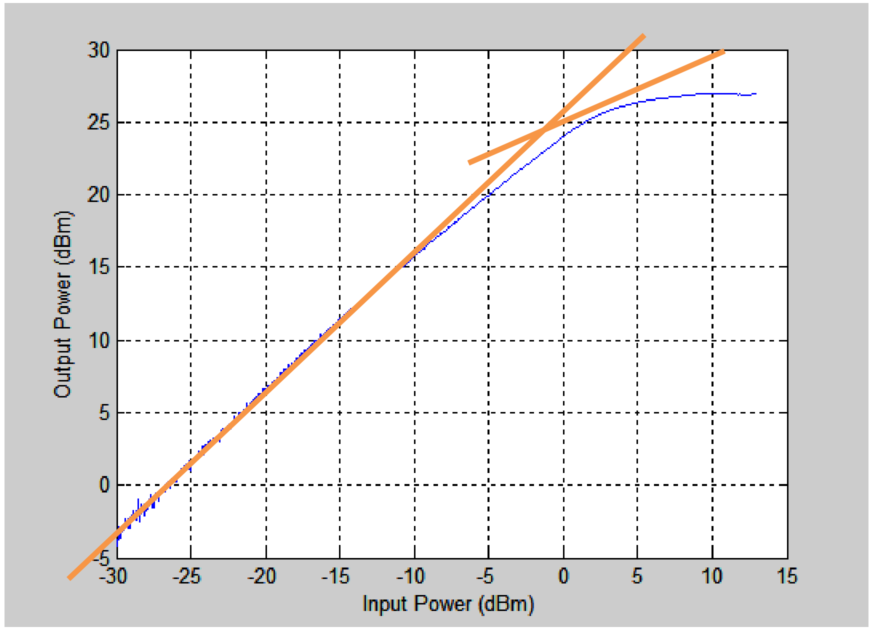
FFT Size: 8 16 32 64 128 [✓ 256]

Close Update Help

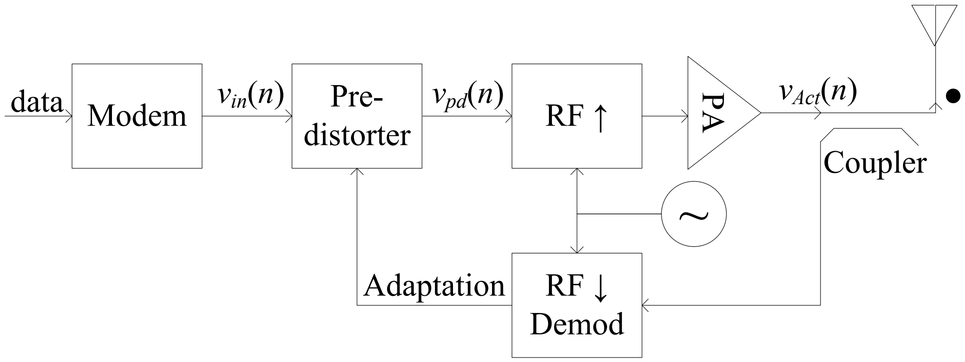
Can also demonstrate coherent sampling

Alter input signal level or clipping level to see change in fundamental and harmonic energy. *Note*: Fundamental gain decreases with input

Measurement of Compression



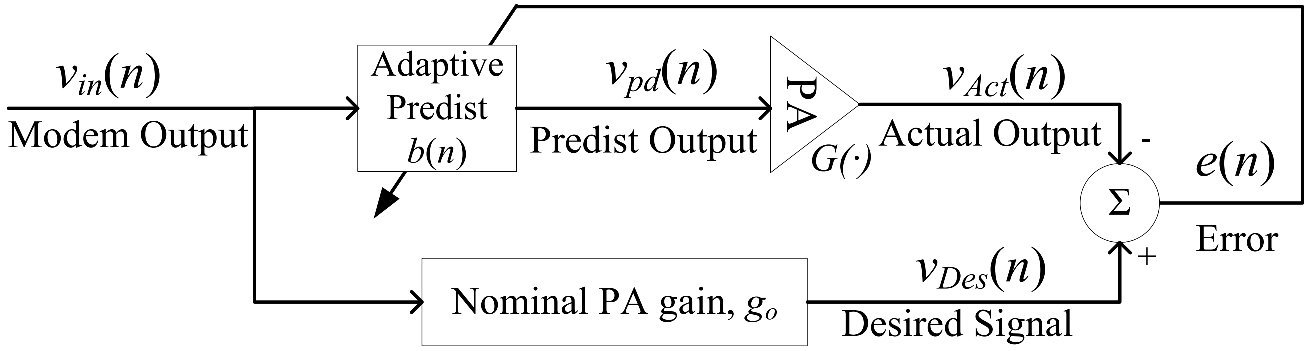
- Ideally gain would be constant.
- Practically gain is compressed near the peak output power of the amplifier
- Logically model amplifier gain as fixed within a bunch of small equal segments of input power called "bins"



- While transmitting data, couple some of the transmitted energy back to the receiver to determine the actual amplifier output.

Gain-Based LUT Predistortion [1]

- For each region, find a constant *b* that such that product of *b* and the actual amplifier gain is equal to the desired gain.
- Initially assume *b* = 1 for each bin.
- For each transmitted point at time *n* within a given bin find a better estimation of the correction factor *b* to use at time (*n*+1) using the LMS algorithm with a learning factor *μ*:



$$e(n) = v_{Des}(n) - v_{Act}(n) = g_o v_{in}(n) - G\big(b(n)v_{in}(n)\big)$$

$$b(n+1) = b(n) + 2\mu e(n) v_{in}^{*}(n)$$

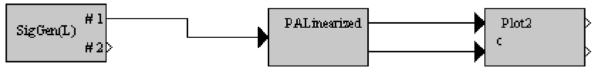
[1] Cavers, J.K., "A linearizing predistorter with fast adaptation," Vehicular Technology Conference, 1990 IEEE 40th , vol., no., pp.41-47, 6-9 May 1990.

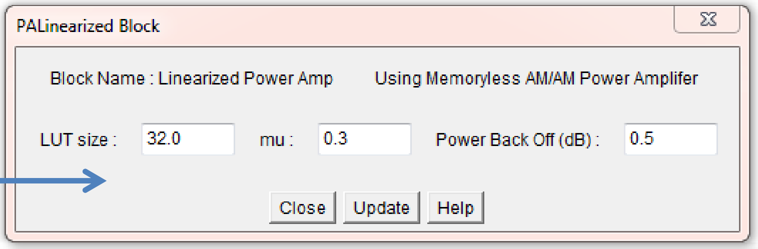
JDSP Simulation: Digital Predistortion

Simulation for 64-carrier BPSK OFDM Signal

When DPD+PA is well linearized, spectral leakage is small.

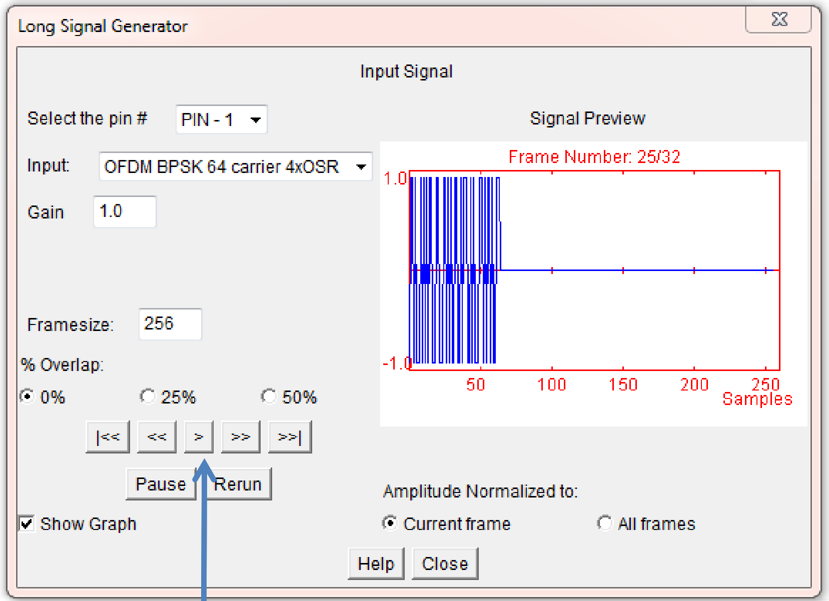
JDSP allows running several OFDM frames through the DPD system with different LUT sizes, Learning Factors, and Power Back off and examining adjacent channel leakage




PALinearized Block

Block Name : Linearized Power Amp   Using Memoryless AM/AM Power Amplifer

LUT size : 32.0   mu : 0.3   Power Back Off (dB) : 0.5


Long Signal Generator

Input Signal

Select the pin # PIN - 1

Input: OFDM BPSK 64 carrier 4xOSR

Gain 1.0

Framesize: 256

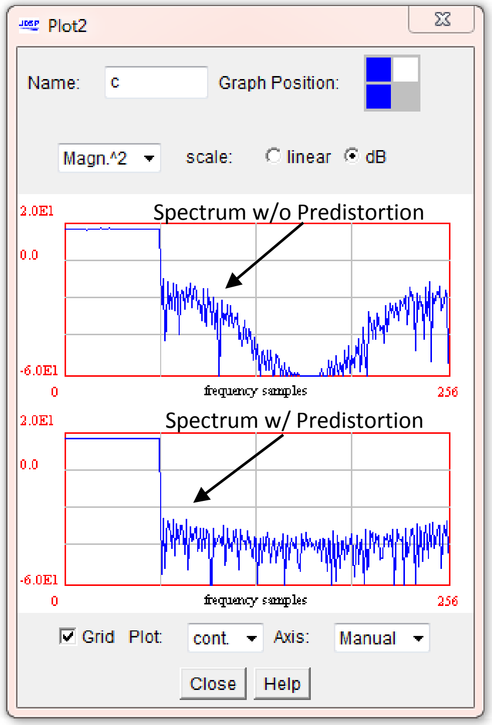
% Overlap: [✓ 0%] 25% 50%

Signal Preview

Frame Number: 25/32

Amplitude Normalized to: [✓ Current frame] All frames

[✓ Show Graph]


Plot2

Name: c   Graph Position:

Magn.^2   scale: linear [✓ dB]

[✓ Grid] Plot: cont. Axis: Manual

Current Frame Control