% Title: Understand and analyze adaptive signal processing.

% Aim: Use LMS algorith to estimate unknown filter transfer function

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% \*\*\*\*\*\*\*\*\*\*\*\*\*\*\*Program starts here\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*

% w(n) digital filter transfer function (adaptive configuration)

% u(n) transfer function

% x(n) input

% e(n) error

% d(n) output

% y(n) output of w(n)

order = 10; % Order of the filter

% Reading .wav file

[x, Fs]=audioread('Lion.wav');

x = x(1:100000); % Consider only 100000 samples

noise = randn(size(x));

nx = x + noise;

nx = nx/max(nx); % Normalizing x

nx = nx';

b = fir1(order, 0.3, 'low'); % filter coeficients of u(n)

d = filter(b, 1, nx); % y(n)

mu = 0.8; % Step size; If mu is very large, it will converge very fast but error may be large. If mu is small, convergence will be slow but, error will be small

lms = dsp.LMSFilter(order+1, 'StepSize', mu, 'WeightsOutputPort', true);

[y, e, w] = step(lms, nx, d);

stem([b.' w]);

title('System Identification by LMS algorithm');

fvtool(b); % Filter visualizatioin of filter coeficients of u(n)

fvtool(w); % Filter visualizatioin of adaptive filter coeficients

% You can observe that adaptive filter will model unknown system almost

% equivalently.

% \*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*Program ends here\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*