Recursive Least Squares Algorithm

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***Abstract*—This manual provides an introduction to the RLS Algorithm.**

1 Source Files

- Get the git source and enter the local directory

```
git clone https://github.com/gadepall/adsp.git/adsp.git
cd adsp/audiosource
```

- Play the **signal_noise.wav** and **noise.wav** files.

2 Problem Formulation

The signal **noise.wav** d(n) contains a human voice along with an instrument sound in the background. This sound is captured in **noise.wav** X(n). The goal is to suppress X(n) in d(n). Let

$$d(n) = e(n) + y(n) \tag{2.0.1}$$

where e(n) is the desired signal. We want an estimate of e(n) from X(n). This can be done by considering

$$y(n) = W^T(n)X(n) \tag{2.0.2}$$

Where

$$X(n) = \begin{bmatrix} X(n) \\ X(n-1) \\ X(n-2) \\ .. \\ .. \\ X(n-M+1) \end{bmatrix}_{MX1} \quad W(n) = \begin{bmatrix} w_1(n) \\ w_2(n) \\ w_3(n) \\ .. \\ .. \\ w_{n-M+1}(n) \end{bmatrix}_{MX1} \tag{2.0.3}$$

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and estimating W(n). The human voice can be characterized as

$$e(n) = d(n)W^T(n)X(n) \tag{2.0.4}$$

The goal is to find W(n) that will allow $W^T(n)X(n)$ to mimic the instrument sound in d(n).