ETIN80 — Algorithms in Signal Processors

Projects

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Projects

Some suggested projects.

- Speech recognition.
- Speech synthesis.
- Speech separation.
- Adaptive line enhancer.
- Adaptive echo canceller.
- Adaptive gain controller.
- Digital communication.
- Beat detection.
- Instrument effects.

Offline vs. Realtime Processing

Offline processing in Matlab has some advantages.

- Non-causal or anti-causal filtering.
- Unlimited memory and processing resources.
- The entire signal is available at all times.

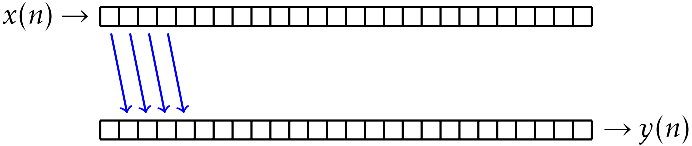
Realtime considerations.

- Limited memory and processing resources.
- The algorithm must run faster than the sample time.
- Sample based processing when delay must be minimised.
- Block processing can reduce the effective processing time.

Block Processing

Process in blocks rather than individual samples.

- Wait for N samples before processing.
- Process all N samples at the same time.
- The supplied framework provides block processing.



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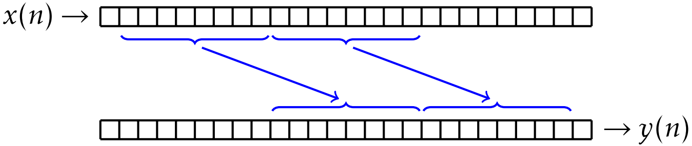


Block Processing

Process in blocks rather than individual samples.

See the function `buffer` in Matlab for block processing.

- `xb = buffer(x, n)`

```
function myproject
    x = audioread('input.wav');

    xb = buffer(x, 320);
    [M, N] = size(xb);
    yb = zeros(M, N);

    for n = 1:N
        yb(:, n) = process(xb(:, n));
    end

    y = yb(:);
end

function y = process(x)
    ...
end
```

Recursive Averaging

Sometimes long-time averaging is required.

- Low memory prevents long buffers for linear averaging.

$$P(n) = \frac{1}{N}\sum_{k=0}^{N-1} x(n-k)^2$$

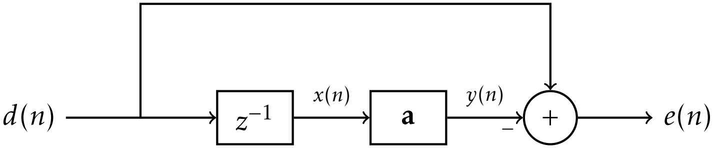
- Recursive averaging allows averaging without memory.

$$P(n) = \alpha P(n-1) + (1-\alpha)x(n)^2$$

Linear Prediction

The 1-step forward linear prediction filter.

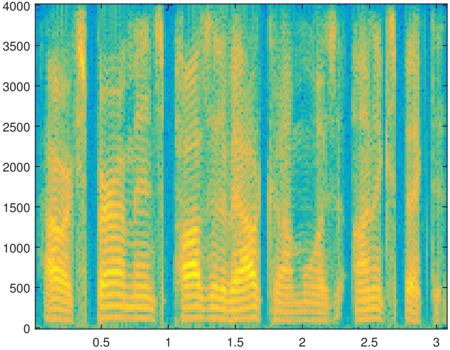
- Wiener problem with analytical or adaptive solutions.
- The filter describes deterministic properties of the signal.
- Common in speech processing.
  - Describes *formants* or *acoustic resonance*.
  - Filter $\mathbf{a}$ is normalized and pitch-independent.



Speech Modeling

Formants and stationarity.

- Formants show up as lines over time.
- Stationary over roughly 20 ms for speech.



Speech Recognition

Recognize spoken words from a pre-defined database.

- Frame the signal into blocks.
- Calculate the prediction filter.
- Compare the LP coefficients to a database.

$$x(n) \longrightarrow \boxed{\text{Pre-emph.}} \longrightarrow \boxed{\text{Framing}} \longrightarrow \boxed{\text{LPC}} \longrightarrow \mathbf{v}_k$$

Speech Synthesis

Transform speech into a robotic voice.

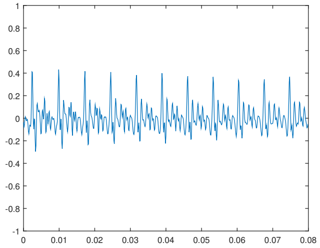
- Frame the signal into blocks.
- Calculate the prediction filter.
- Calculate the pitch from the error signal.
- Construct a new excitation signal for different effects.
- Filter with the inverse of the prediction filter.

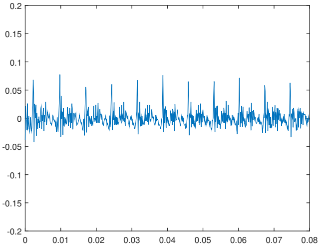
$$x(n) \longrightarrow \boxed{\mathbf{a}} \longrightarrow e(n) \qquad\qquad e'(n) \longrightarrow \boxed{\mathbf{a}^{-1}} \longrightarrow x'(n)$$

Speech Synthesis

Linear prediction analysis.

Input signal and error signal.

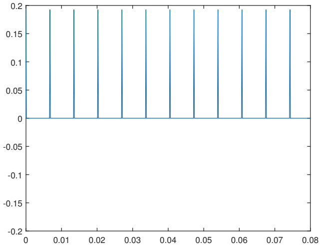


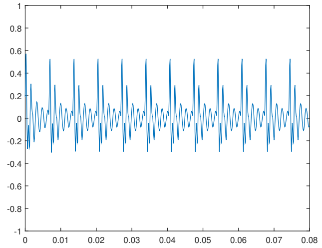


Speech Synthesis

Linear prediction synthesis.

Synthetic error signal and reconstructed output signal.





Speech Separation

Separate two simultaneous speech sources.

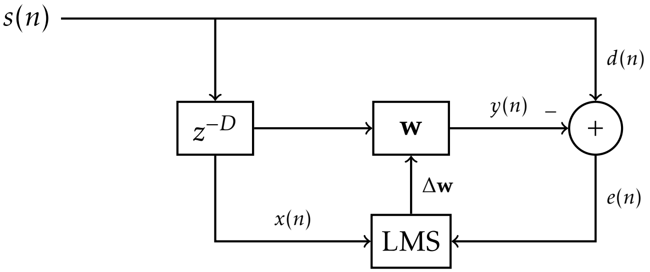
- Speech excite narrow frequency bands at short times.
- Different speech sources rarely overlap.
- Separate different sources with selective masking.

$$Y_n(\omega, \tau) = \begin{cases} X(\omega, \tau) & \text{if source } n \text{ is active in } (\omega, \tau), \\ 0 & \text{otherwise.} \end{cases}$$

Adaptive Line Enhancer

Remove tonal components from a signal.

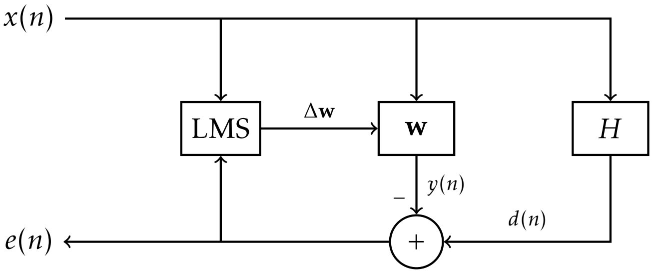
- Delays a signal and attempt to predict it.
- Extract uncorrelated components such as speech.
- Suppress correlated components such as tonal sounds.



Adaptive Echo Cancellation

Remove echo form a signal.

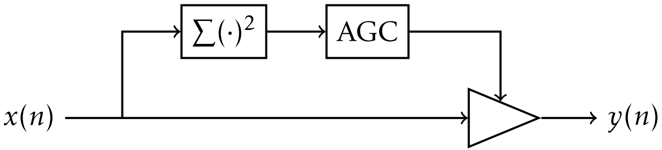
- Identify the far-end to near-end channel.
- Extract near-end speech.
- Suppress far-end speech or disturbance.



Adaptive Gain Controller

Adjusts a gain to compress or expand the signal.

- Suppress high signal levels.
- Pass normal signal levels.
- Suppress background noise levels.



Digital Communication

Transmit digital information using modulation.

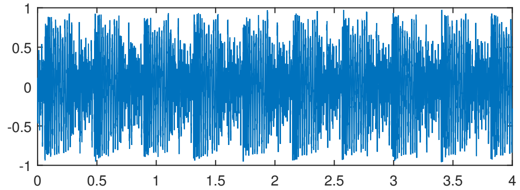
- Implement a transmitter and a receiver structure.
- Decide on modulation method and parameters:
  - PSK, FSK, UWB modulation format.
  - Communication protocol.
  - Message length and synchronization.
  - Error correction.

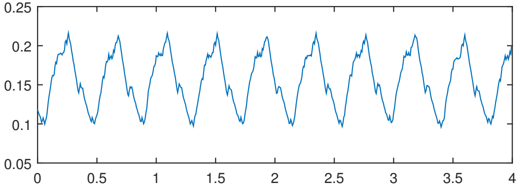
$$x(n) \longrightarrow \boxed{\mathrm{TX}} \longrightarrow y(n) \rightsquigarrow y'(n) \longrightarrow \boxed{\mathrm{RX}} \longrightarrow \hat{x}(n)$$

Beat Detection

Find beats in music.

- Beats in a music dictates its rhythm.
- Mixing of music relies on beat matching.





Instrument Effects

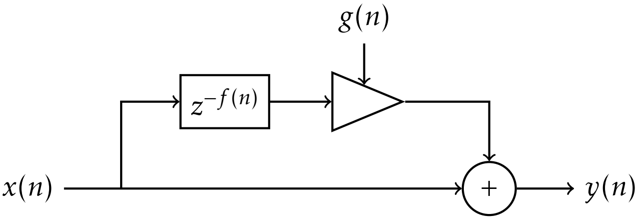
Bank of audio effects for instruments.

Time based effects.

- Delay, echo, reverb.

Modulation based effects.

- Chorus, flanger, ring modulator, tremolo.



Your own project

Or choose your own project...

- Base on your own personal interest or field of research.
- Project details and focus can be adapted as necessary.