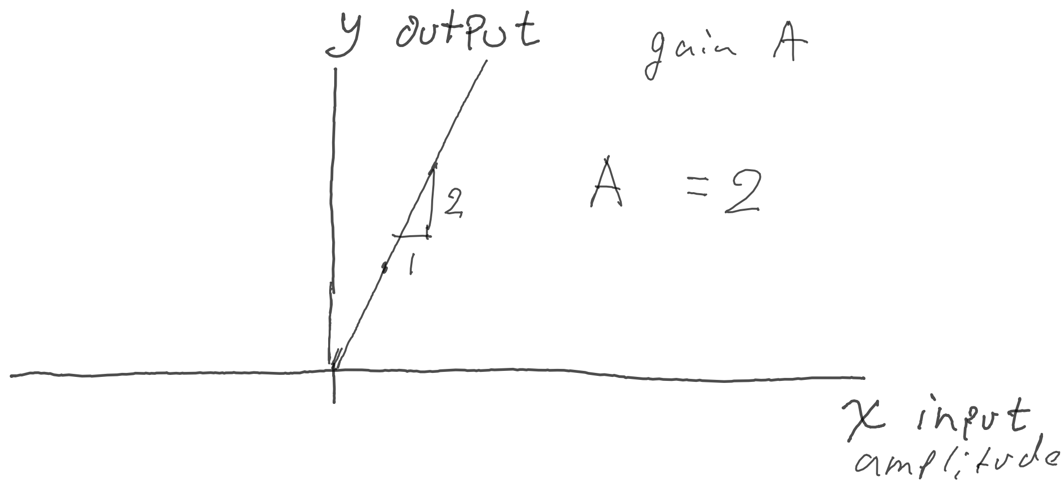
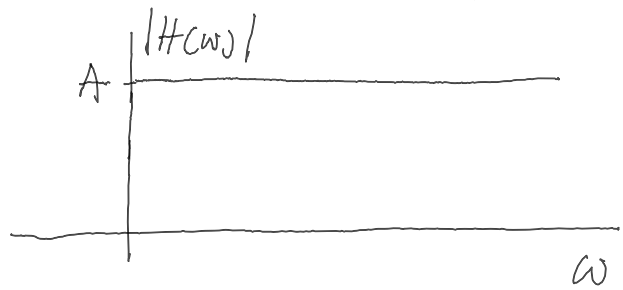


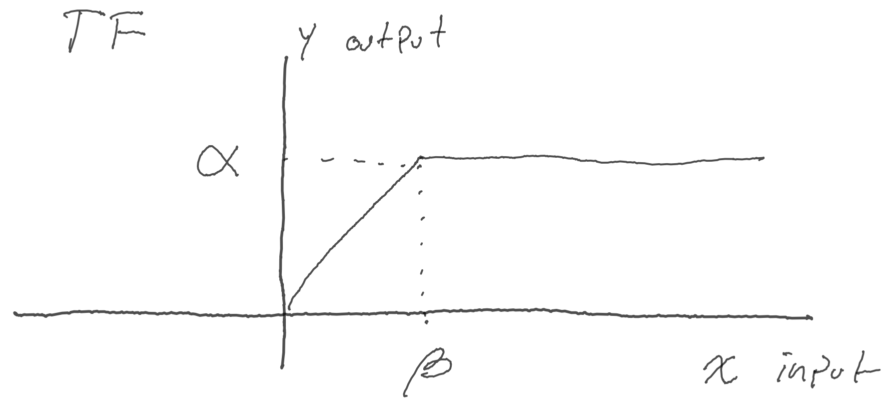

- For an input amplitude of 1
  You get an output of amplitude 2

  ↳ does not depend on frequency

AKA
flat Freq Response

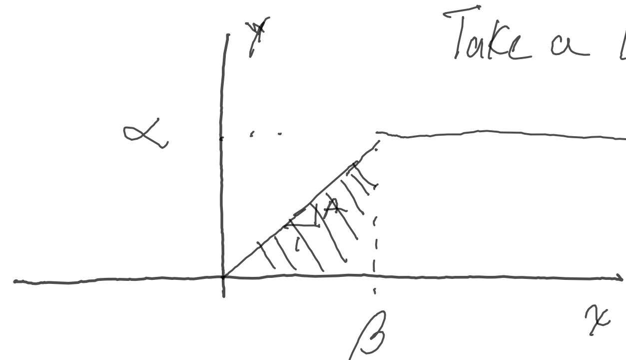


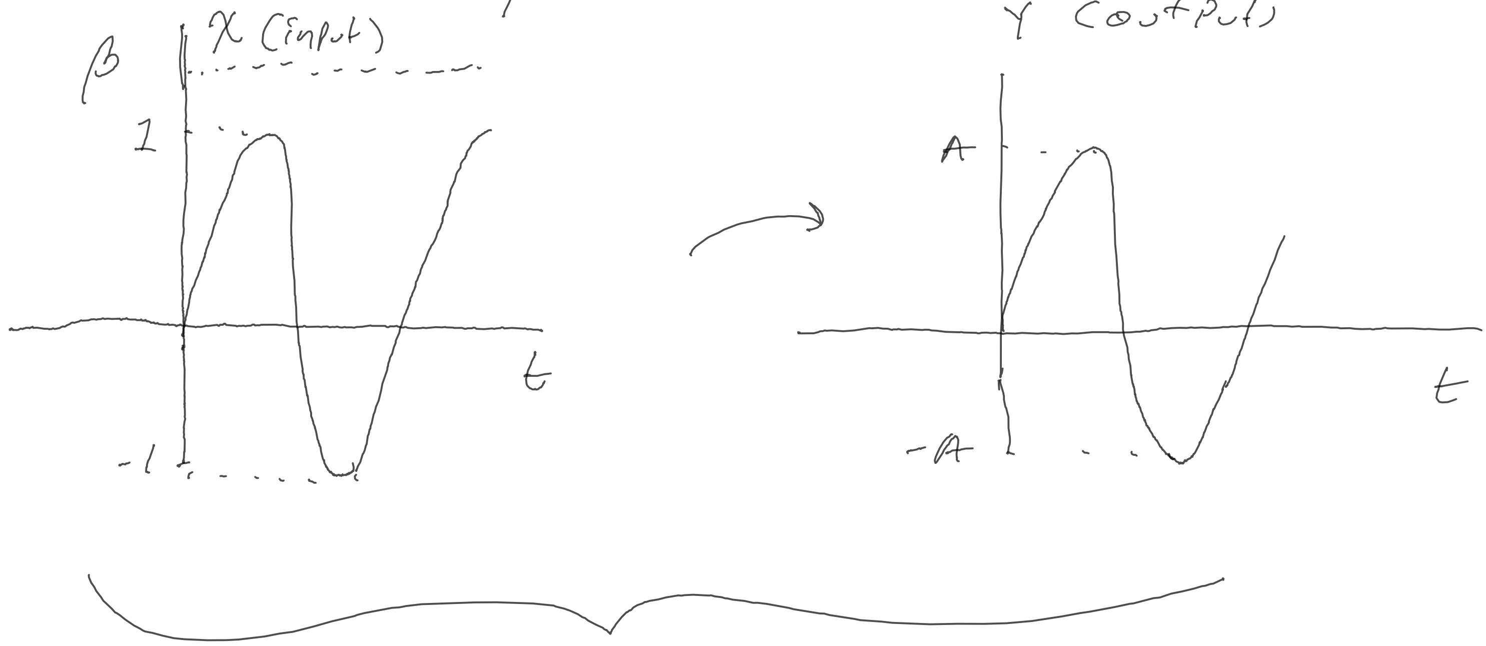

Take non-linear TF




- Constant gain A until input amplitude exceeds β
  then the output forces to α. This is a limiter
  Clips at $x = \beta$

Take a look at the linear Region



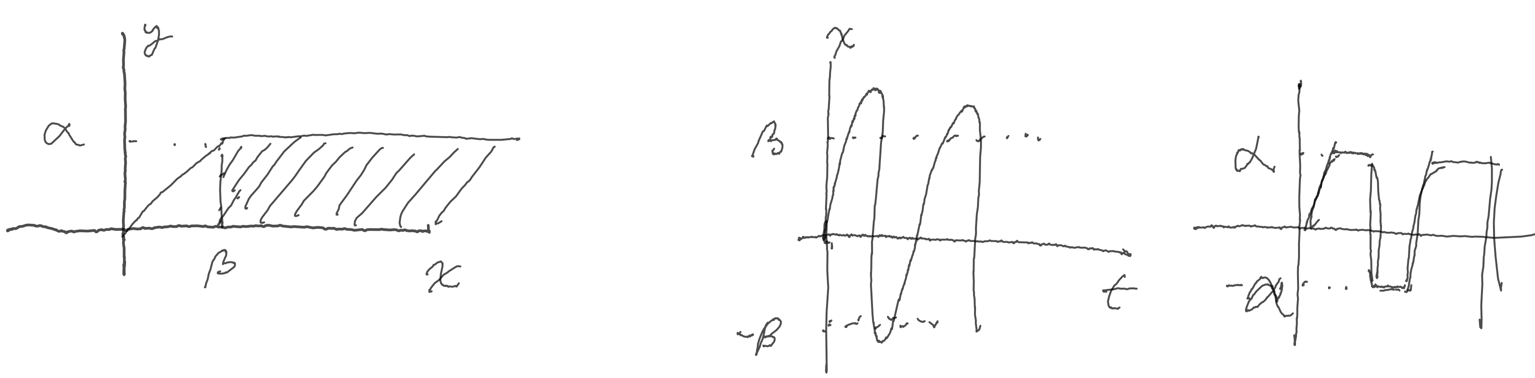




This is in the constain gain linear region.

The output is simply scaled by gain of system.

- Does not depend on frequency at all

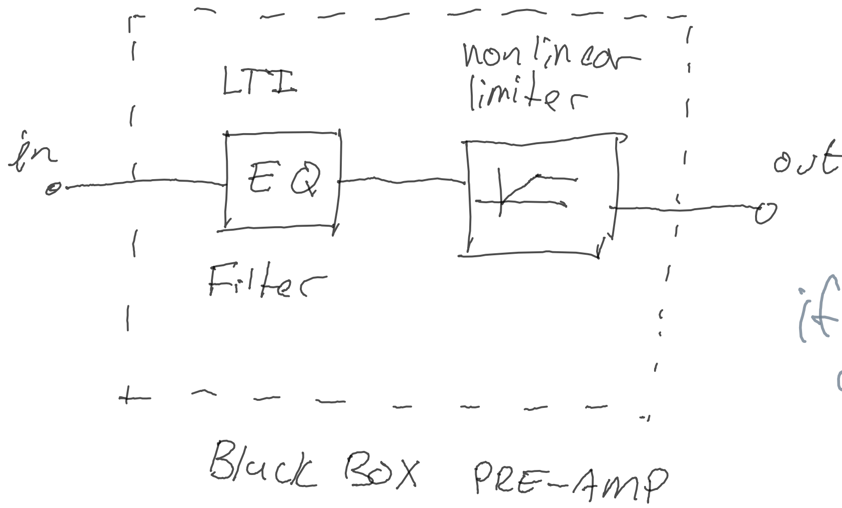
## NON-LINEAR REGION



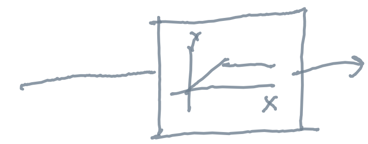

Still does not depend on frequency.

- Wave shaper type system.

Imagine the Pre amp is an EQ followed by a waveshaper.




if modeled as



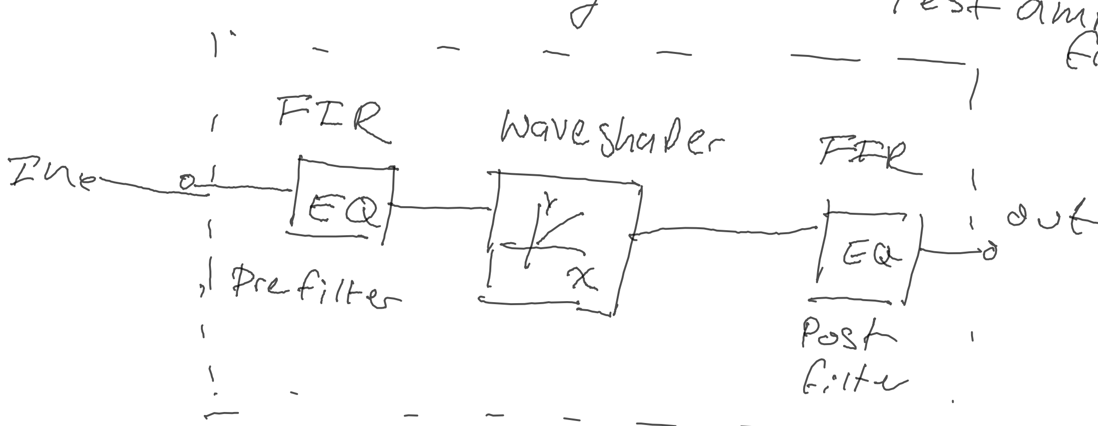

Now there is some frequency dependent stuff goin on. How do we model this?

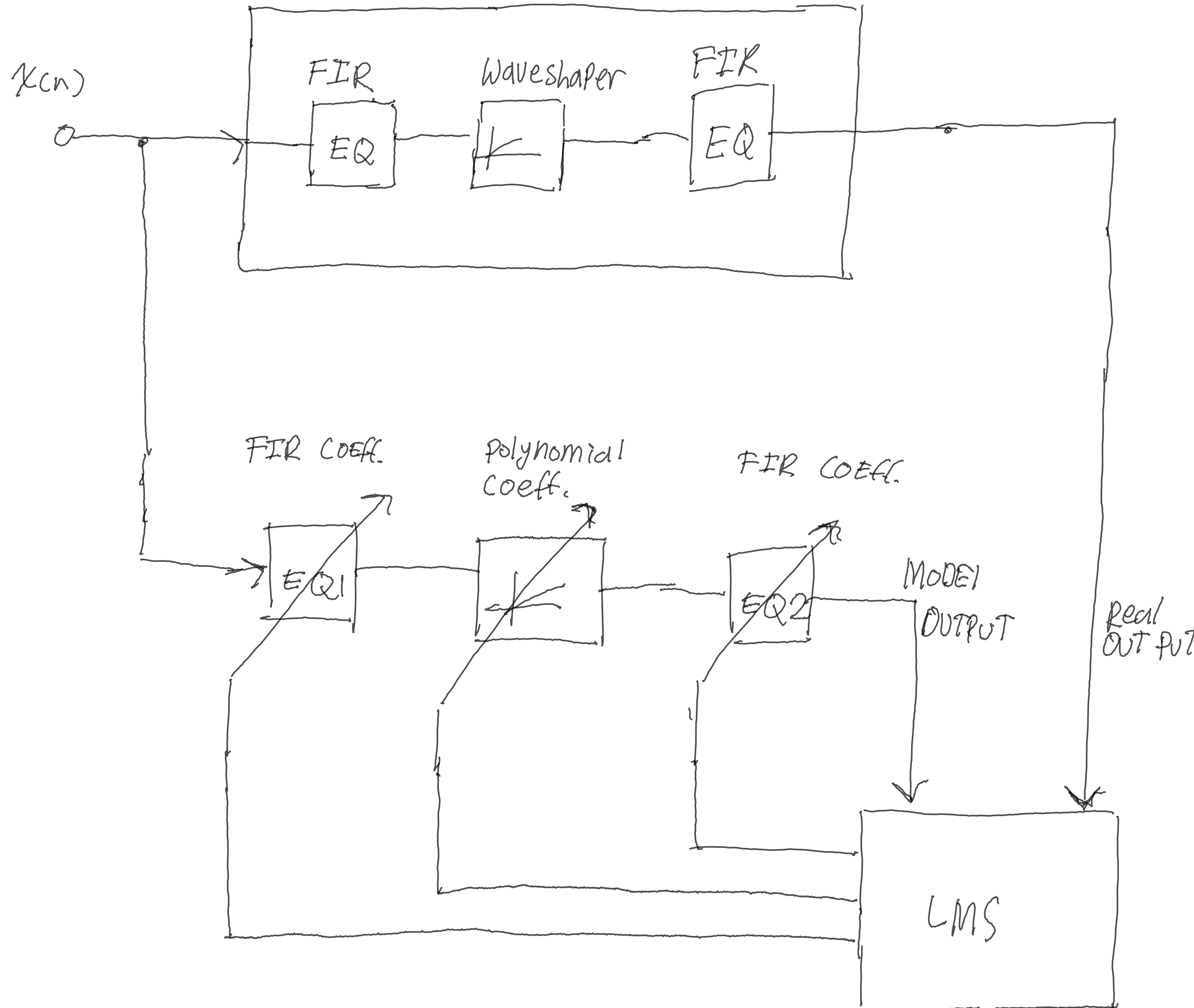
There could be 2 signals with same amplitude and different frequency and the limiter model breaks down.

Basically if the Preamp has any filtering whatsoever then we need a more advanced model.

Perhaps something like:

Need to iteratively test amplitude alongside frequency.

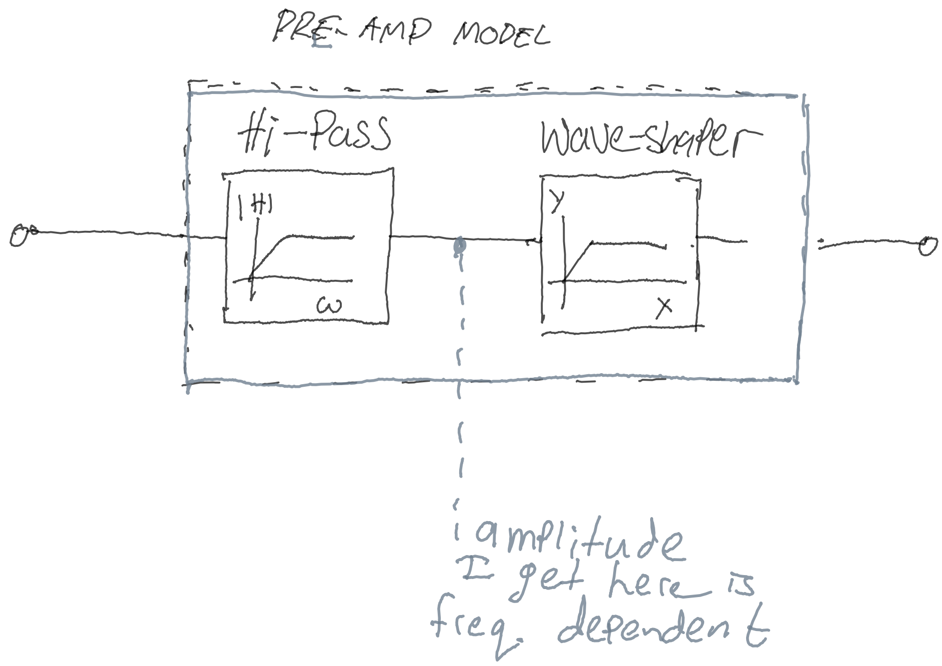





- How to nudge each coefficient each iteration?
- What are my test inputs going to be?

→ could do some genetic approach to sort of randomly at first select coefficients then eventually get more and more accurate

Assume all inputs + outputs are bounded by 1




need to define the system for both frequency Response and amplitude response.

ex. For an input with frequency $\alpha$ and amplitude A the output has frequency $\beta$ and amplitude B

I am Confused

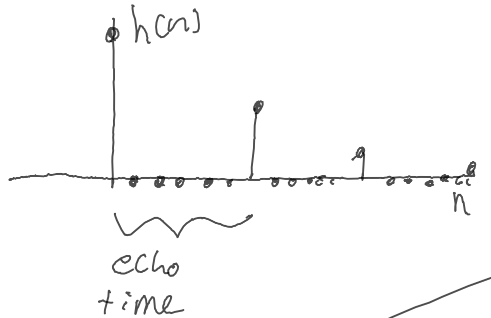
Non-linear is hard

# Questions for Cheever

## REVERB MODEL

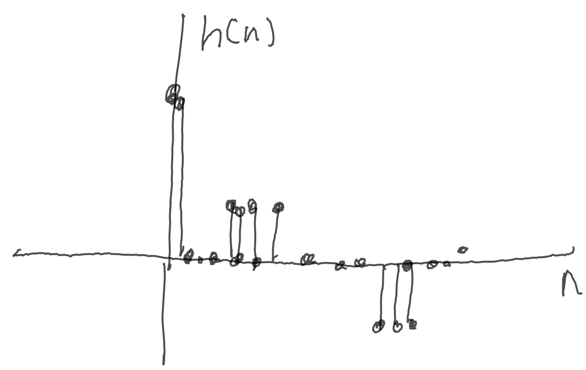
- Old tarble reverb isn't just a series of echoes

like




it has both pos + negative impulse like:

once plotted the IR in MATLAB




What does the negative sign mean? is it that there are 180 degree flipped echoes?

↳ or is it something to do with how it was recorded with the microphone → some high pass that would knock off any DC perhaps.

## NON-LINEAR SYSTEMS

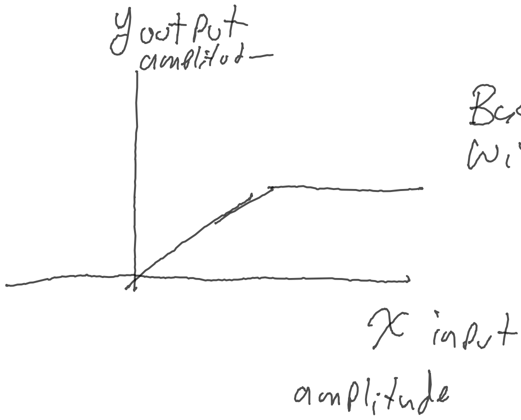
- where do I even begin with these?
  - are freq response measurements completely useless now?
  - Modeling a system with input/output meassurements?

What Does a nonlinear system model even mean?

- It's not a freq filter anymore so we are not after FIR coefficients.

- What values/coeff. are we searching for? When we want to characterize a syste?

↳ or does this depend (as it usually does)?




Basic transfer characteristic model with some polynomial coefficients (Chebyshev Polynomials)

$$y(x) = c_0 x + c_1 x^2 \ldots$$

adaptively (LMS) find coeff $C_k$

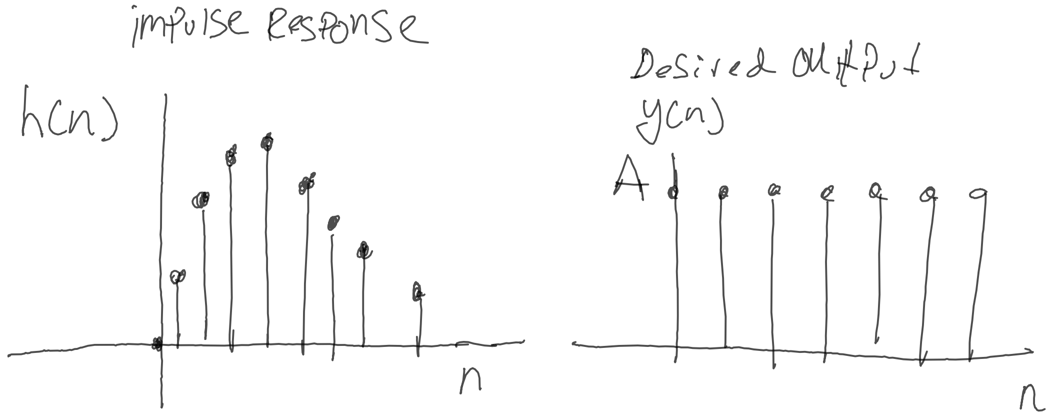
$\boxed{\text{CAFFEINE CONVOLUTION}}$

Say impulse Response is


blood caffeine level after espresso shot at t=0

Say a user wants a flat caffeine level what is the input to achieve this?

I thought this was a simple "deconvolution"




how to find $x(n)$ such that

$$x(n) * h(n) = y(n)$$

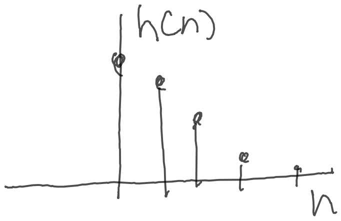
$$X(z)\,H(z) = Y(z) \Longrightarrow \boxed{X(z) = \frac{Y(z)}{H(z)}}$$

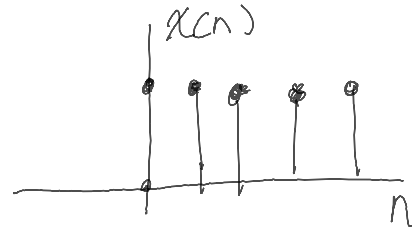
how do I do this numerically in Matlab?

What does inverse convolution look like in time domain? integral wise.

WEIRD OP AMP BEHAVIOR

Convolution as a som of scaled impulse Responses



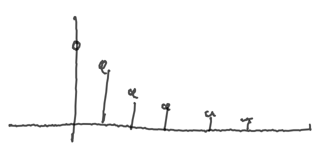




$$x(n) = \sum_{K=-\infty}^{K=\infty} x(K)\,\delta(n-K)$$

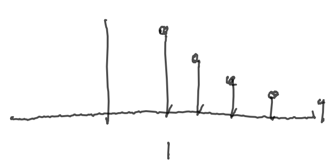
$\Downarrow$

$$y(n) = \sum_{K=-\infty}^{K=\infty} x(K)\,h(n-K)$$

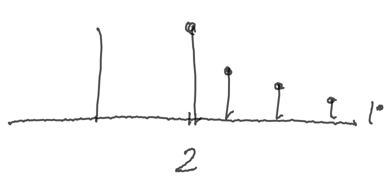
$h(n-0) = h(n)$



$h(n-1)$




$h(n-2) =$



$$y(n) = \sum_{k=-\infty}^{\infty} x(k)\, h(n-k) = \sum_{k=-\infty}^{\infty} x(n-k)\, h(k)$$

How to show this change of variables?

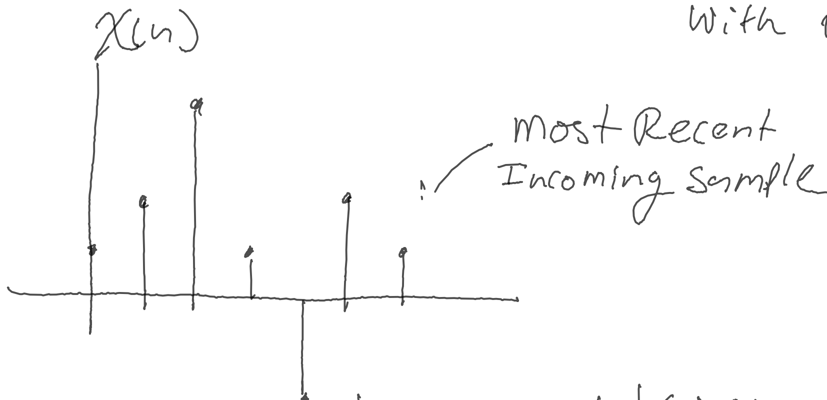
$$x(n) * h(n) = h(n) * x(n)$$

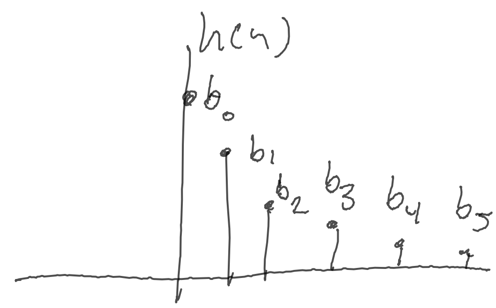
REAL TIME FORM (h(k) is known before and new x(n) coming in)

$$y(n) = \sum_{k=-\infty}^{M} x(n-k)\, h(k)$$

next output only depends on past inputs

In this form the next output sample to compute is the sum of the past M samples with the h(n) FIR coeff.




$$y(n) = h(0)\,x(n) + h(1)\,x(n-1) + h(2)\,x(n-2) + \ldots$$

$$y(n) = b_0\, x(n) + b_1\, x(n-1) + b_2\, x(n-2) + \ldots$$

Most Recent sample multiplied w/ $b_0$ (first sample of h(n))

①

$$y(n) = \sum_{k=-\infty}^{\infty} x(k)\, h(n-k)$$

$h(n)$ shifted to right by $k$

Can be thought of as simply time shifted impulse responses, scaled by the amplitude of $x(k)$

- This interpretation makes sense in non-real time where we already have the entire $x(n)$ set.

THIS IS THE INTUITIVE WAY TO LOOK AT IT

②

$$y(n) = \sum_{k=-\infty}^{\infty} x(n-k)\, h(k) \quad \text{or} \quad \sum_{k=0}^{M} x(n-k)\, h(k)$$

is useful when thinking about <u>Real time applications.</u>

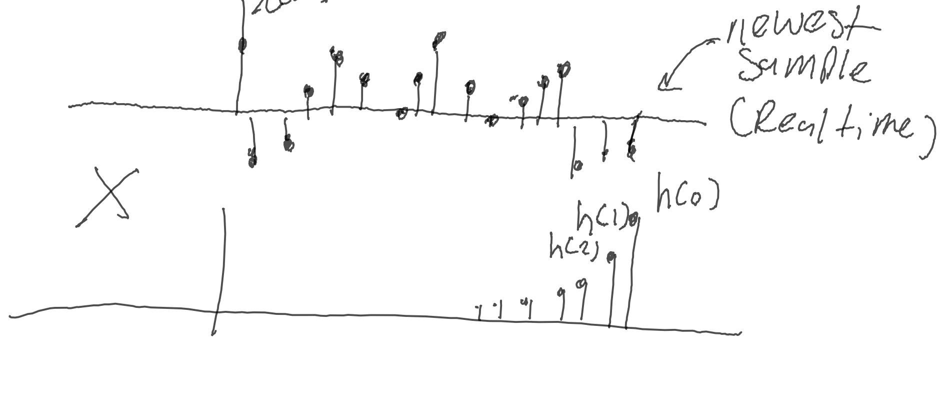
$$y(n) = x(n)h(0) + x(n-1)h(1) + x(n-2)h(2) + \ldots$$

In this form its more obvious that ② is true.

this is like taking the previous M inputs and each multiplying by FIR coefficients

↳ This way its more obvious to see the $h(n)$ flipped across the y axis and sliding over 1 to the right each time a new sample comes in

This is the USEFUL WAY TO THINK ABOUT IT



# LINEAR Systems

- We define a system w/ its transfer function $H(s)$ or $H(z)$

$$H(s) = \frac{Y(s)}{X(s)}$$

$$\underbrace{Y(\omega)}_{\text{spectrum of output}} = \underbrace{H(\omega)}_{\text{system response}} \underbrace{X(\omega)}_{\text{spectrum of input}}$$

↳ in my mind I sort of equate $H(z)$ w/ $H(s)$ or $H(z)$ with $H(\omega)$ by assuming

all I need is a frequency response to compute the output for any input.

↳ But obviously $H(\omega)$ is just $H(z)$ evaluated along the unit circle. This is just a slice of the whole picture. So what useful info does the rest of $H(z)$ give us?

- all signals are sums of sinusoids so maybe freq response is all that is needed??

Why don't we evaluate the TF over different paths other than the unit circle?

SO All I need to define a system (Filter) are a set of accurate enough filter coefficients.

Here's what I think about my question
on System classification + linear Systems

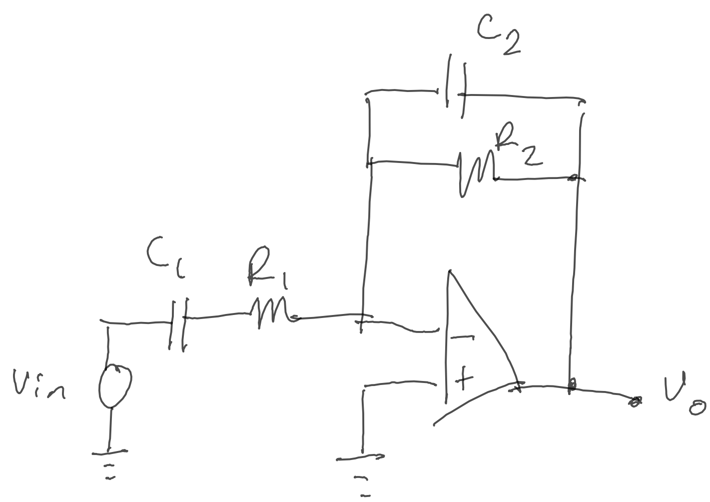
- goes back to the Laplace domain

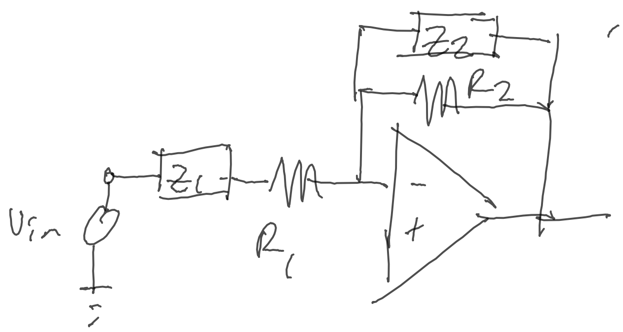
We learned the laplace transform to turn
a linear differential eqn (system) to an algebraic
expression so we could solve it more easilly.

$$\ddot{y} + 5\dot{y} + 6y = 2\dot{x} + 1$$

ignoring initial
Conditions we get
the homogeneous Part

$Y(s)s^2 + 5$



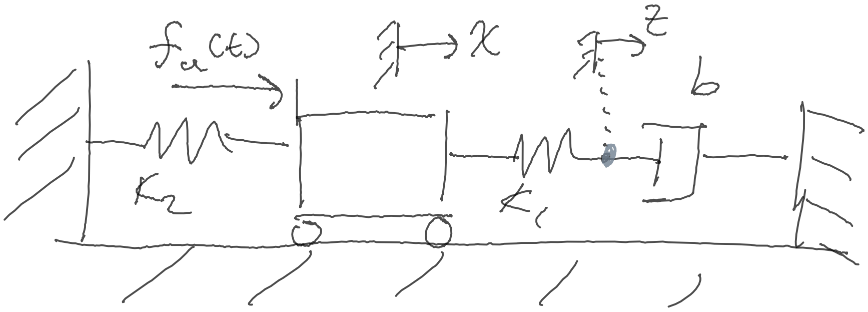




$$f_a - k_1(z-x) - xk_2 = m\ddot{x}$$

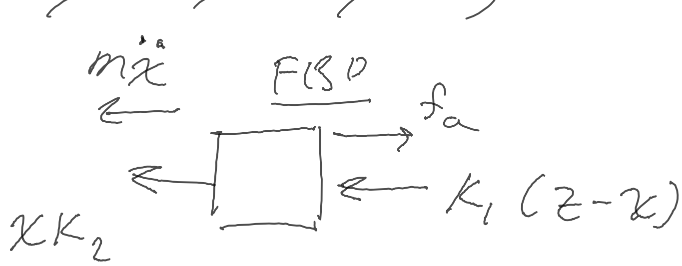
$$\frac{V_o}{V_{in}} = \frac{-Z_2 \| R_2}{Z_1 + R_1} \quad \begin{matrix}(1)\\(2)\end{matrix}$$

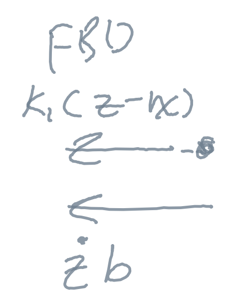
$$m\ddot{x} + k_1(z-x) + xk_2 = f_a$$

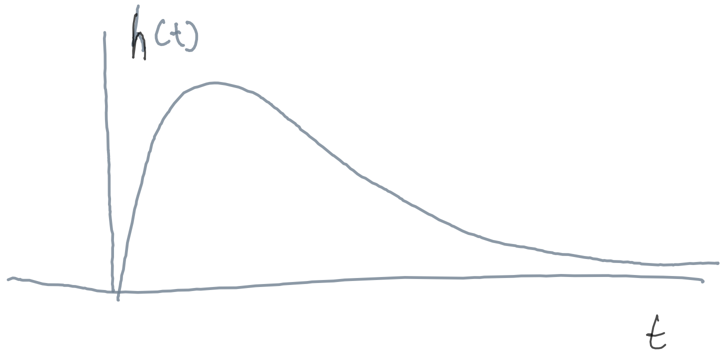
$$F_{net} = ma \qquad \sum F = ma$$



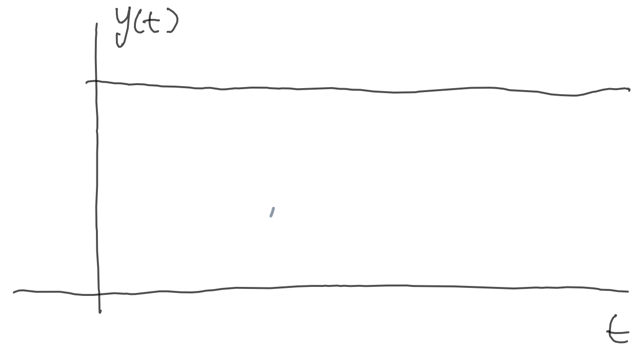

$$k_1 z - k_1 x = \dot{z} b$$











Want an $x(t)$ such that $x(t) * h(t) = y(t)$

$$X(s) \cdot H(s) = Y(s)$$

$$X(s) = \frac{Y(s)}{H(s)}$$

---

Say we have som TF $H(s)$ and find its $h(t)$ impulse response experimentally to be $h(t)$.

Why is it ok to take $H(\omega)$ as the system when that is only a slice of $H(s)$?

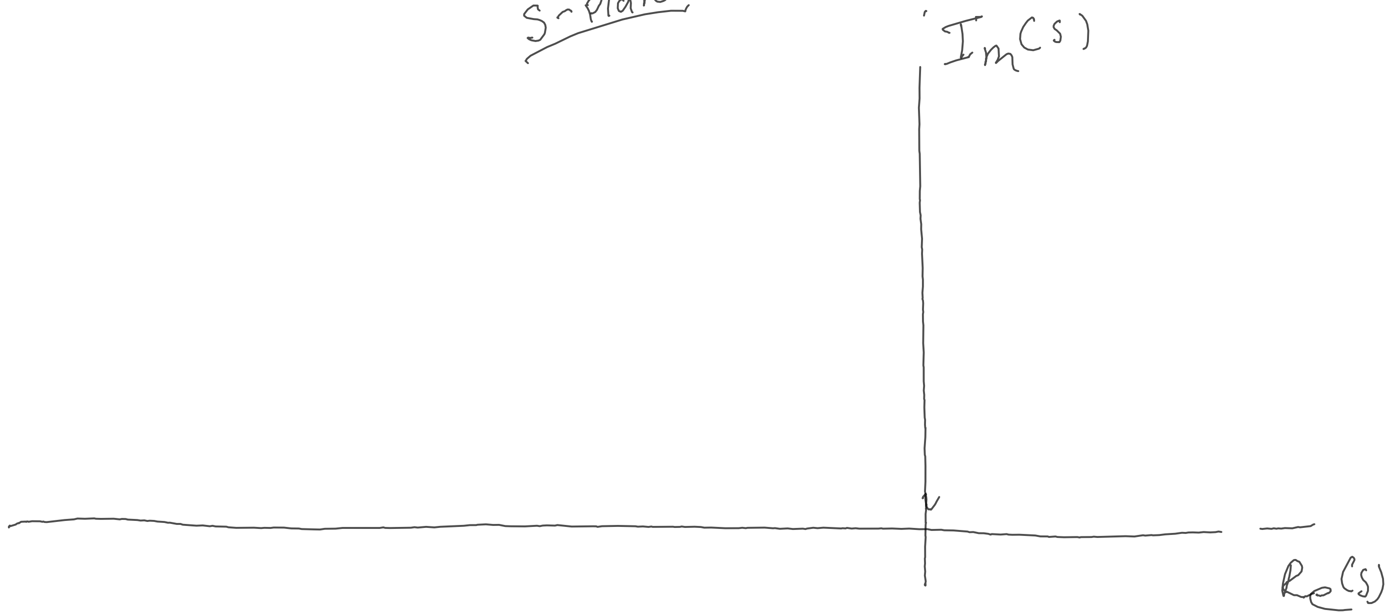
Why does

$$x(t) * h(t) = y(t)$$

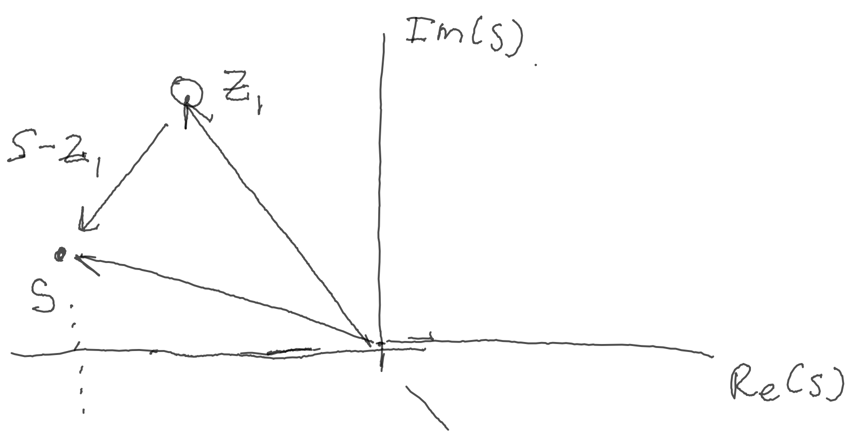
$$X(s) H(s) = Y(s)$$

$$X(s) = \frac{Y(s)}{H(s)} \xrightarrow{\mathcal{F}} X(\omega) = \frac{Y(\omega)}{H(\omega)}$$

S-plane

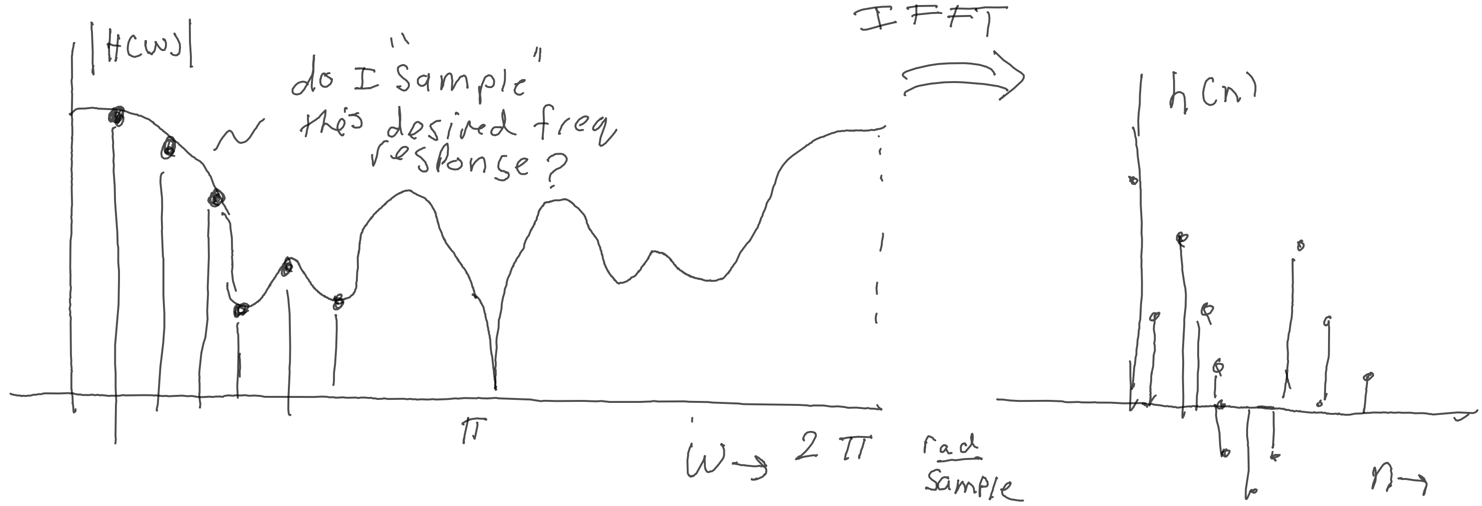



$$H(s) = K\,\frac{(s-z_1)(s-z_2)(s-z_{m-1})(s-z_m)}{(s-p_1)(s-p_2)(s-p_{n-1})(s-p_n)}$$




S is the particular point in the S plane at which the TF is to be evaluated

$(s - z_1)$ represents a vector pointing from $z_1$ to S, the point at which the function is evaluated




Computational method of designing arbitrary filters?,

↳ take Inverse DTFT to get h(n)

use that h(n) as FIR COEFF?

then that filter will have a cont. f.R. closely matches to the desired.

---

a measured impulse Response can only get you to find the system frequency Response, not the full transfer function,

- Freq Response is just a slice of the full picture.

- Using imp. responce / measured freq response is not enough to classify full system or is it?

  - I guess all signals are sums of sinusoids.

- Math aside - how is $\mathcal{L}(\delta(t)) = 1$ ?

FIR FILTERS are

finite in length and thus their DTFTS are also finite length and discrete: ..

freqz doing to make its frequency response look continuous given a set of coefficients?

-maybe zero padding it?

DTFT is always continuous function of $\omega$

$$X(\omega) = \sum_{n=0}^{M-1} x(n) e^{-j\omega n}$$

no matter how many M coefficients

$X(\omega)$ is a continuous function of $\omega$.

~~Datasheet Spacing~~

~~Vcap Application notes~~

~~Vref Application notes~~

PMOD Pinout