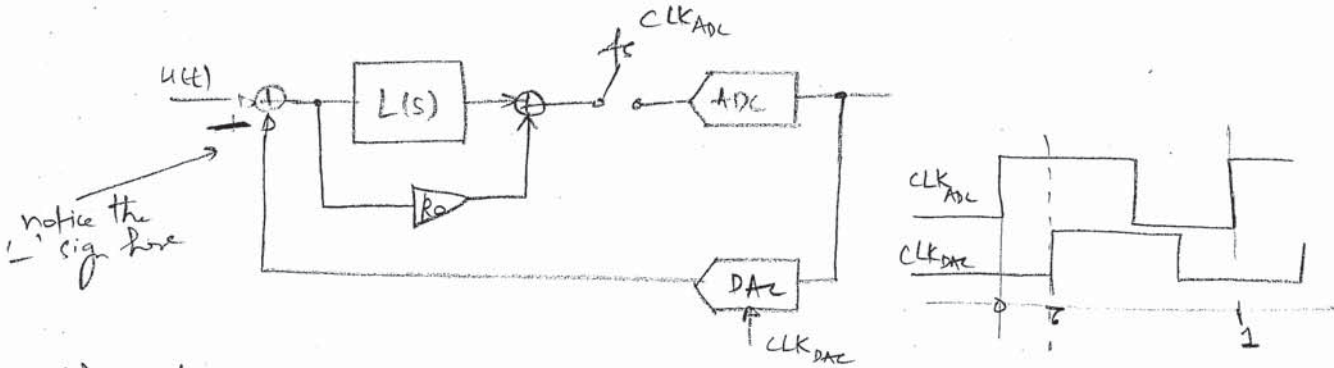


③ Direct feedback path around the quantizer



$$\Rightarrow L'(s) = k_0 + \frac{k_1}{s} + \frac{k_2}{s^2}$$

we had $L(z) = \frac{-2z+1}{(z-1)^2}$

Extra feedback path provides the extra control parameters in the loop response.

Now, if when the ELD is completely compensated:

$$\Rightarrow k_0 z^{-1} + k_1 \times (\text{RHS of (A)}) + k_2 \times (\text{RHS of (B)}) = \frac{-2z+1}{(z-1)^2} = \frac{2z-1}{(z-1)^2}$$

going through the algebra we get:

$$\left. \begin{aligned} 0.5z^2 k_2 - zk_1 + k_0 &= 0 \\ (0.5 - z + 0.5z^2)k_2 + (1-z)k_1 + k_0 &= 2 \\ -(0.5 + z - z^2)k_2 + (1-2z)k_1 + 2k_0 &= 1 \end{aligned} \right\} \rightarrow \text{①}$$

Solving this set of equations we get.

$$\{k_0', k_1', k_2'\} = \{1.5z + 0.5z^2, 1.5 + z, 1\}$$

Verify for $z=0$, $\{k_0', k_1', k_2'\} = \{0, 1.5, 1\}$ ← Same as expected.

$\Rightarrow k_1$ is tuned and k_0 is added.

↳ This process requires one to go back and forth s- and z- domains,
 ↳ tedious algebra
 ↳ painful for higher-order modulators.

Pavan's solution for loop filter comprising of ideal integrators (no resonators).

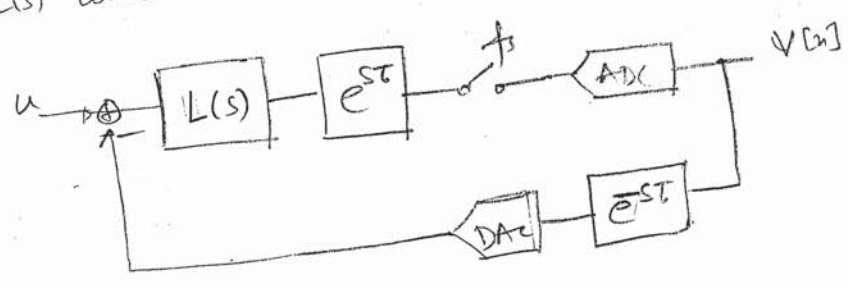
The technique:

if the CT ΔΣ loop filter is

$L(s) = \frac{k_1}{s} + \frac{k_2}{s^2} + \dots + \frac{k_n}{s^n}$ with no exom-loop delay.

What value of coefficients $\{k_1, k_2, \dots, k_n\}$ must be chosen so that the NTF remains same even when there is an ELD of T .

Conceptually we can compensate for ELD by cascading $L(s)$ with a block with TF e^{sT}



clearly $e^{sT} \xleftrightarrow{\mathcal{L}^{-1}} s(t+T)$

is non-causal.
↳ not realizable in practice.

We get an interesting case when, the input is piecewise constant and we are only interested in the sampled output
↳ e^{sT} can be reinterpreted to obtain

$H(s) = H(s) e^{sT} = k_1 \frac{e^{sT}}{s} + k_2 \frac{e^{sT}}{s^2} + \dots + k_n \frac{e^{sT}}{s^n}$

A. NRZ DAC (Proof later)

(9)

for NRZ DAC, expand e^{sT} as the polynomial is 's', such that

↳ for i^{th} integrator $\frac{e^{sT}}{s^i}$ is truncated beyond the i^{th} power of s

⇒

$$\frac{e^{sT}}{s} \rightarrow \frac{1}{s}(1+sT) = \frac{1}{s} + T \rightarrow \textcircled{1}$$

$$\frac{e^{sT}}{s^2} \rightarrow \frac{1}{s^2}(1+sT + \frac{(sT)^2}{2}) = \frac{1}{s^2} + \frac{T}{s} + \frac{T^2}{2} \rightarrow \textcircled{2}$$

$$\frac{e^{sT}}{s^i} \rightarrow \frac{1}{s^i} \left(1 + \dots + \frac{s^i T^i}{i!} \right) = \frac{1}{s^2} + \frac{T}{s^{i-1}} + \dots + \frac{T^i}{i!} \rightarrow \textcircled{N}$$

Then, the loop filter TF whose samples are identical to $L(z)$, are given by the weighted summation of the RHTs of these \textcircled{N} equations by k_1, \dots, k_N resp.

$$\Rightarrow \hat{L}(s) = k_1 \left(\frac{e^{sT}}{s} \text{ exp.} \right) + k_2 \left(\frac{e^{sT}}{s^2} \text{ exp.} \right) + \dots + k_i \left(\frac{e^{sT}}{s^i} \text{ exp.} \right) + \dots + \frac{k_N}{s^N}$$

$$\hat{L}(s) = \left(k_1 T + k_2 \frac{T^2}{2} + \dots + k_N \frac{T^N}{N!} \right) + \frac{(k_1 + k_2 T + \dots + k_N \frac{T^{N-1}}{N-1!})}{s} + \dots$$

$$+ \dots + \frac{k_i + k_{i+1} T + \dots + k_N \frac{T^{N-i}}{N-i!}}{s^i} + \dots + \frac{k_N}{s^N}$$

$$\Rightarrow \left\{ \begin{aligned} k_0' &= k_1 T + \frac{k_2 T^2}{2} + \dots + k_N \frac{T^N}{N!} = \sum_{i=1}^N k_i \frac{T^i}{i!} \\ k_1' &= k_1 + k_2 T + \dots + k_N \frac{T^{N-1}}{N-1!} = \sum_{i=1}^N k_i \frac{T^{i-1}}{i-1!} \\ &\vdots \\ k_N' &= k_N \end{aligned} \right.$$

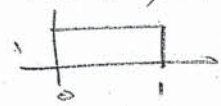
∴ verify the result for the second order case by using $\{k_1, k_2\} = \{1, 5, 13\}$

Proof:

$u_k(t) \rightarrow$ unid step function integrated 'k' times

$u_0(t) = u(t) \leftarrow$ unid step function 

The DAC pulse is $u(t) - u(t-1)$, initially assume $z=0$



$x_i(t) \leftarrow$ output of the i^{th} integrator.

$N \leftarrow$ order of the loop filter

$y(t) \leftarrow$ output " " " "

\Rightarrow we have

$$y(t) = \sum_{i=1}^N k_i x_i(t) = \sum_{i=1}^N k_i (u_i(t) - u_i(t-1))$$

the sampled output

$$y[n] = \sum_{i=1}^N k_i x_i[n] = \sum_{i=1}^N k_i (u_i[n] - u_i[n-1])$$

when the DAC pulse is delayed by 'z',

the integrator and loop filter outputs become $x_i(t-z)$ and $y(t-z)$.

Using Taylor series for $0 < z < 1$, we have the ideal sampled output of the i^{th} integrator can be expressed as:

$$x_i[n] = x_i(t) \Big|_{t=n} = x_i(t-z) \Big|_{t=n} + z \frac{dx_i(t-z)}{dt} \Big|_{t=n} + \dots + \frac{z^l}{l!} \frac{d^l x_i(t-z)}{dt^l} \Big|_{t=n} + \dots$$

\hookrightarrow ①

Now, since,

$$\frac{dx_i(t-z)}{dt} = x_{i-1}(t-z) \Big|_{t=n} = 0$$

① reduces to

$$x_i[n] = \left[x_i(t-z) + z x_{i-1}(t-z) + \dots + \frac{z^l}{l!} x_0(t-z) \right] \Big|_{t=n}$$

$$f(t) = f(t_0) + \frac{f'(t_0)}{1!} (t-t_0) + \frac{f''(t_0)}{2!} (t-t_0)^2 + \dots + \frac{f^{(l)}(t_0)}{l!} (t-t_0)^l + \dots$$

$$f(t) = x_i(t)$$

with $t = n$ i.e. here

$$t_0 = t - \tau \Big|_{t=n}$$

$$\Rightarrow x_i(t) = x_i(t-\tau) + \frac{x_i'(t-\tau)}{1} \tau + \dots + \frac{x_i^{(l)}(t-\tau)}{l!} \tau^l + \dots$$

$$x_i(t) \Big|_{t=n} = x_i(t-\tau) \Big|_{t=n} + \frac{\tau}{1} x_i'(t-\tau) \Big|_{t=n} + \dots + \frac{\tau^l}{l!} x_i^{(l)}(t-\tau) \Big|_{t=n} + \dots$$

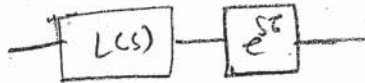
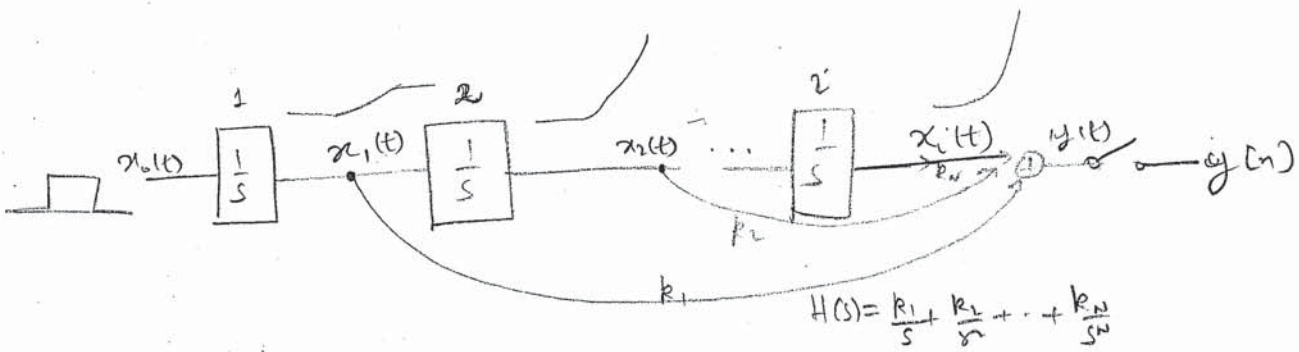
ideal integrator output
current integrator outputs

$$x_i'(t-\tau) = x_{i-1}$$

$$\underline{x_0(t-\tau) = 0}$$

$$\Rightarrow x_i[n] = \left[x_i(t-\tau) + \tau \cdot x_{i-1}(t-\tau) + \dots + \frac{\tau^l}{l!} x_0(t-\tau) \right] \Big|_{t=n}$$

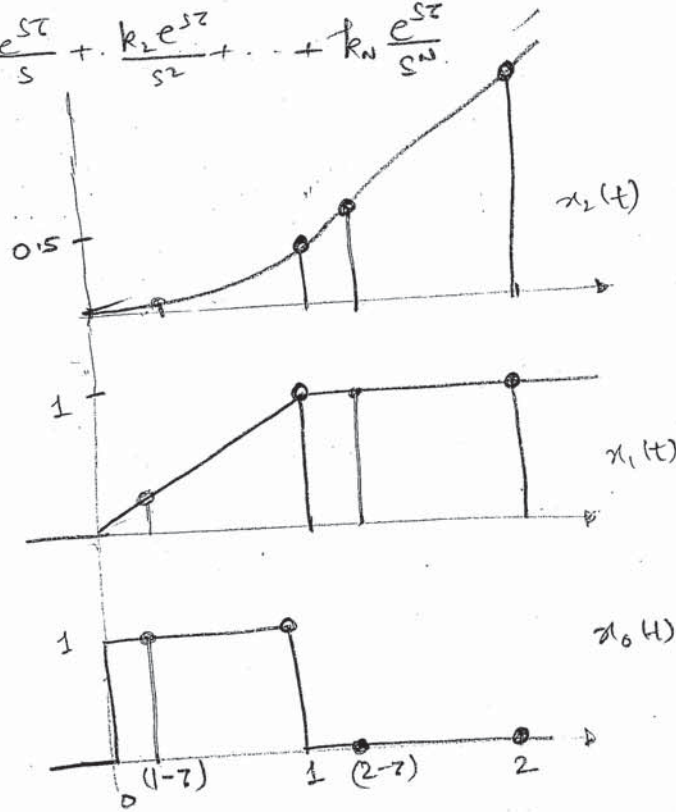
$$\frac{1}{s^i} + \tau \frac{s}{s^i} + \dots + \frac{\tau^l}{l!} \frac{s^l}{s^i} = \frac{1}{s^i} \cdot e^{s\tau} \Big|_{(i+1)\text{ term}}$$



$$\hat{L}(s) = L(s)e^{sT} = k_1 \frac{e^{sT}}{s} + \frac{k_2 e^{s2T}}{s^2} + \dots + \frac{k_n e^{snT}}{s^n}$$

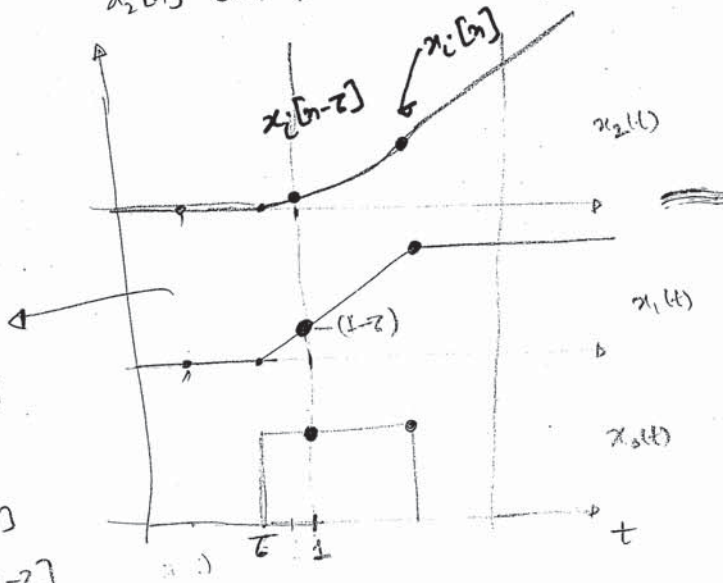
$$x_0(t) = u(t) - u(t-1)$$

$x_1(t)$



\Rightarrow for a second-order modulator, and NRZ DAC,

$x_2[n]$ can be determined from $x_2[n-2]$, $x_1[n-2]$ and $x_0[n-2]$.



reds can be linearly combined to estimate green dots.

$$x_1[n] = x_1[n-2] + z \cdot x_0[n-2]$$

$$x_2[n] = x_2[n-2] + z x_1[n-2] + \frac{z^2}{2} x_0[n-2]$$

⇒ Even with delayed DAC pulse, the ideal output samples of the i^{th} integrator can be generated by combining the output samples of the i^{th} integrator and the preceding $(i-1)$ integrators as well as the input to the loop filter $x(t-z)$. (11)

In frequency-domain, we can say that the ^{ideal} output samples of the i^{th} integrator, can be obtained by sampling the output of a filter whose transfer function is given by

$$\frac{1}{s^i} + z \frac{s}{s^i} + \dots + \frac{1}{L^i} z^i \frac{s^i}{s^i} = \frac{1}{s^i} e^{sT} \Big|_{i+1 \text{ terms}}$$

⇒ The TF of the compensated loop filter is given by

$$\begin{aligned} \hat{L}(s) &= \frac{k_N}{s^N} \left(1 + sT + \frac{s^2 T^2}{L} + \dots + \frac{s^N T^N}{L^N} \right) \\ &+ \dots + \frac{k_i}{s^i} \left(1 + sT + \frac{s^2 T^2}{L} + \dots + \frac{s^i T^i}{L^i} \right) \quad \rightarrow (2) \\ &+ \dots + \frac{k_1}{s} (1 + sT) \end{aligned}$$

⇒ from (2) the direct path comes out as a direct consequence

$$k_0' = k_1 + k_2 \frac{T}{L} + \dots + k_N \frac{T^N}{L^N}$$

B) RZ feedback DAC (proof omitted).

Two cases:

(i) $z < 0.5$, the order of the system doesn't increase

LELD compensation done by coeff tuning.
Same as NRZ derivation, but the expansion of e^{sT} in $\frac{e^{sT}}{s^i}$ is truncated after $(i-1)^{th}$ power of s .

$$\Rightarrow \frac{e^{sT}}{s^i} \rightarrow \frac{1}{s^i} \left(1 + \dots + \frac{(sT)^{i-1}}{(i-1)!} \right) = \frac{1}{s^i} + \frac{T}{s^{i-1}} + \dots + \frac{T^{i-1}}{s \cdot (i-1)!}$$

\Rightarrow the uncompensated loop TF is

$$G(s) = \frac{k_1 + k_2 z + \dots + k_N \frac{z^{N-1}}{N-1}}{s} + \frac{k_i + k_{i+1} z + \dots + k_N \frac{z^{N-i}}{(N-i)!} + \dots + \frac{k_N}{s^N}}$$

(ii) $z > 0.5$, a direct path is necessary in addition to coefficient tuning.
 \hookrightarrow see ref for details.

\rightarrow

\Rightarrow simple band-pass all is s-domain!

\hookrightarrow method requires that the higher order derivatives of the i^{th} integrated output become 0 when driven by a piecewise constant DAC pulse.

\hookrightarrow Not true when STF with complex zeros is used.

\hookrightarrow pulse responses now contain sine and cosine.

\hookrightarrow no easy solution exist !!

\hookrightarrow The low pass formulae do an acceptable job of stabilizing the loop-delay for large OSR, low-pass DSMs.

\hookrightarrow No solution for BP-DSMs.

Issues with Table based methods :

↳ (S+) Certainly need mathematical analysis for better understanding/modeling of the system.

↳ The algebra (for the general case) is tedious and unwieldy

↳ TF of real op-amp have several poles/zeros due to finite A_{OL}, fin effects. (assuming Active-RC implementation)

↳ obtaining these polezero locations not an easy task!

↳ the system may not have a solution when the integrators are non-ideal and the poles of L(z) are different from the poles of the integrator paths

eg for integrators with finite gain, poles of real loop filter will not be at z=1.
⇒ can not be solved.

Numerical fitting approach

↳ implemented in the realizeNTF.ct function.

$l[n]$ ← column vectors of N samples

eg. for $NTF(z) = (1-z)^2$, we have

$$l[n] = [0 \ 2 \ 3 \ \dots \]^T$$

• The colⁿ vectors formed by N samples of the pulse responses of the direct path and the integrator outputs are denoted as

$$l_0[n] = [0 \ 1 \ 0 \ \dots \ 0]^T$$

$$l_1[n] = [0 \ (1-z) \ 1 \ \dots \ 1]^T$$

$$l_2[n] = [0 \ 0.5(1-z)^2 \ (1.5-z) \ \dots \ (N-0.5-z)]^T$$

Choose ' N ' such that it is much larger than the number of unknowns to be determined. Then we have the weighting coefficient $K = [k_0 \ k_1 \ k_2]^T$, determined by solving.

$$[l_0[n] \ l_1[n] \ l_2[n]] K = l[n]$$

$$\begin{matrix}
 \begin{bmatrix}
 0 & 0 & 0 \\
 1 & 1-z & 0.5(1-z)^2 \\
 0 & 1 & 1.5-z \\
 \vdots & \vdots & \vdots \\
 0 & 1 & N-0.5z
 \end{bmatrix}
 \begin{bmatrix}
 k_0 \\
 k_1 \\
 k_2 \\
 \vdots \\
 0
 \end{bmatrix}
 =
 \begin{bmatrix}
 0 \\
 2 \\
 3 \\
 4 \\
 \vdots \\
 N+1
 \end{bmatrix}
 \\
 \begin{matrix}
 N \times N & N \times 1 & N \times 1
 \end{matrix}
 \end{matrix}$$

more equations than unknown with ideal integrators, the above set of equations admit a unique solution. \Rightarrow K is independent of N .

↳ does away with tedious Algebra
↳ easily obtained from simulation

↳ Has issues when real opamps are used
↳ read paper by Shanthi.