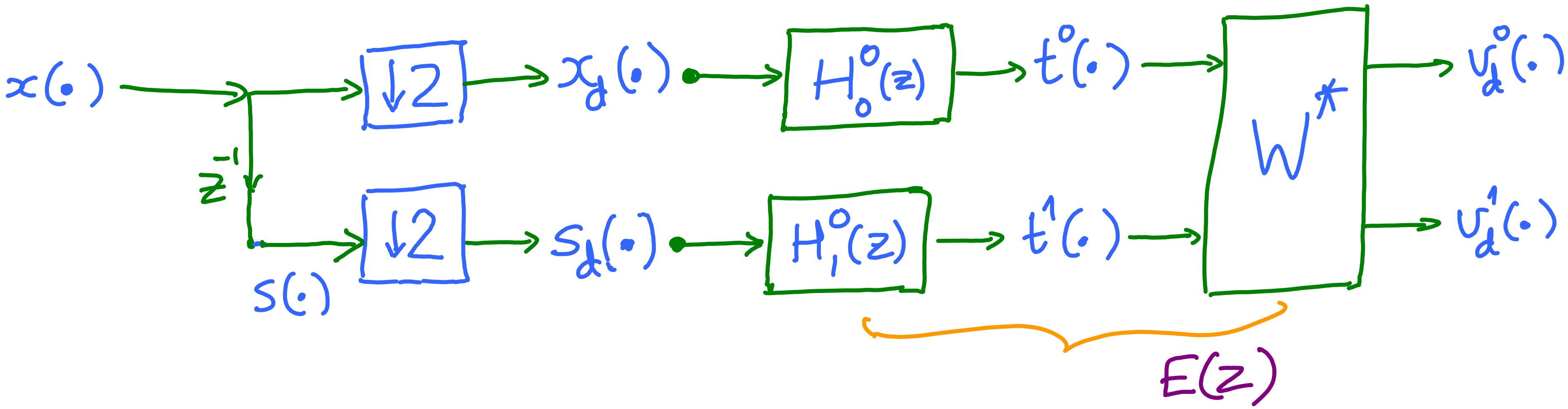
EE6133 Experiment-2

Expt-2a:

In this experiment, you will implement a 2-channel DFT-FB in polyphase form. The analysis FB has the structure



Note that $W^* = W = \begin{bmatrix} 1 & 1 \\ 1 & -1 \end{bmatrix}$.

$H_0^0(z)$ and $H_1^0(z)$ are the 2-polyphase components of the prototype $H^0(z)$.

$H^0(z)$ should be designed as an equiripple Type-2 Linear-Phase LPF, with $\omega_p = 0.45\pi$ and $\omega_s = 0.55\pi$. It is causal with support $[0, N]$.

After designing $H^0(z)$, identify $H_0^0(z)$ and $H_1^0(z)$ for use in the analysis and synthesis FBs.

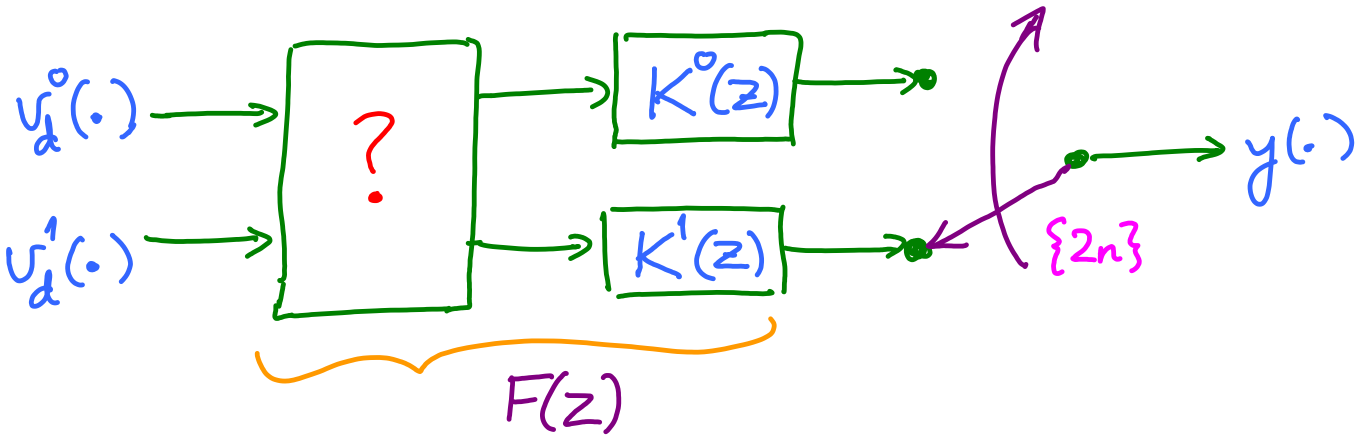
Assume that the input $x(\cdot)$ starts with $x(-1)$.

$s(0) = x(-1)$

$\Rightarrow$ The starting samples of $s(\cdot)$ and $s_d(\cdot)$ are $s(0)$ and $s_d(0)$.

$\Rightarrow$ All downstream signals $t^0(\cdot)$, $t^1(\cdot)$, $v_d^0(\cdot)$, $v_d^1(\cdot)$ start with index=0.

The synthesis FB has the structure discussed in Lecture-18:



Connect the outputs of the analysis FB to the inputs of the synthesis FB.

Let $K^0(z)$ and $K^1(z)$ be causal filters whose support starts at 0.

Then $v_d^0(\cdot)$, $v_d^1(\cdot)$ start with index 0 $\Rightarrow$ output $y(\cdot)$ also starts at 0.

You should position the synthesis commutator accordingly at the start of synthesis.

Implement the following choices for $K^0(z)$ and $K^1(z)$:

1. $K^0(z) = H_1^0(z)$ and $K^1(z) = H_0^0(z)$ ($\Rightarrow$ No aliasing, but not PR)
   Plot the resulting $T_{zp}(\omega)$ (as derived in Lecture-18) on a linear scale for $0 \leq \omega \leq \pi$.

2. $K^0(z) = H_0^0(z)$ and $K^1(z) = H_1^0(z)$
   This is an intentionally bad choice, which results in aliasing.

Determine the FB output for each of these 2 choices, for each of the 2 given input clips.

Deliverables for Experiment-2a:

- Value of N & Magnitude response of $H^{o}(z)$
- Plot of $T_{zp}(\omega)$ for the FB without aliasing
- Magnitude spectrum of the 2 audio output signals for both synthesis configurations (ie spectrum of all 4 output signals)
- Verbal comparison of the 2 output signals for the same input based on
  (a) Audio Quality and (b) Magnitude Spectrum
- Source Code: Your code must show the analysis & synthesis FBs as separate modules;
  The analysis FB must be the same for Choices 1 & 2 specified above
- All the above deliverables in one report (for each student)

Expt-2b:

In this experiment, you will implement the CMFB analysis & synthesis flowcharts as specified in the MP3 spec.

- Implement the flowcharts exactly as given, with output $\{S_k\}_{k=0}^{31}$ of analysis FB = input $\{S_i\}_{i=0}^{31}$ of synthesis FB
- The array $\{D[i]\}_{i=0}^{511}$ in the synthesis spec is at Line 480 of https://github.com/FlorisCreyf/mp3-decoder/blob/master/tables.h ; obtain the analysis array as $\{C[i] = D[i]/32\}$
- Run the "16kHz music" audio input through the CMFB; compare the output audio with input. As in Expt-2a, also compare the output spectrum with input spectrum.
- Set $\theta_k = 0 \; \forall k$ in both the analysis & synthesis FB, with no other change; run the above expt. Compare this audio output with the previous output. Also compare the spectra of the two outputs & report your observations.

[End]