By pass

16 QAM → ↑4 → PS Filter → Upconversion

↓

Down conversion → Matched Filter → ↓4 → 16 QAM

Above is what we build for D1

16 QAM symbol comes from mapper circuit, upsample by 4, Upconversion upsamples and shifts center frequency of transmitter

Receiver downconversion shifts back down to baseband and downsample

Both of these blocks would cancel each other out irl so we wont build for D1

we compare 16 QAM signals from output and compare it with 16 QAM input

D1 has 3 main circuits: do in this order (p.102)
1) MAtched filter
2) Pulse shaping filter
3) Pulse shaping filter (no multipliers)

Multipliers are an expensive resource. want to make a less power hungry, etc design

Steps to follow:
1. Choose coeffficents
2. Choose signal formats (input, coeff, output, intermediate wires, etc)
3. Choose filter structure
4. Code & test filter

PS and matched are design as SRRC filters, when we cascade these 2 filters, their convolution is a Raised cosine filter

Tb in 456 refers to symbol period (Tsym) ch9 456 notes

"beta = 1 has roll off immediately"

Nyquist theorem: IR in time domain is 0 @ every multiple of Tb

perfect RC has an infinitely long impulse response but this is not practical

slide 24 Ideal SRRC & RC responses (digital)

To go from analog to digital, we need to choose a sampling rate where we can evaluate the response of the SR(t) at multiples of taht sampling rate.

We choose our sampling rate to be 4 times the symbol rate

t = Tb/4 will give us the last expression on pg.24

cascading filters in frequency domain multiples their Frequency response

parameter L refers to Nsps

SRRC IR doesn't have periodic zero crossings like the RC's IR

center coeff is n=0, we truncate the response from -10 to 10 and exclude the rest of the coeffs. However to implement a filter, we cant have coeffs with a negative value since this corresponds to a non-causal filter where we require knowledge of the filter before it happens (not possible); thus we take the IR and shift the response up by so its causal

**Above is Receiver filter

**Transmitter filter/Pulse shaping filter**

We need to design our filter to meet the required stopband and MER; stopband is from 0.2 to 0.5 cycles/sample, Mag Resp of filter MUST be 40 dB below its DC Mag Resp

.5 cycles/samples corresponds to pi rads/sample

on slide 24, our b = 0.25 and L = 4; but in practice it wont be ideal like in the equation for the FR of an ideal SRRC

when we implement in fixed point logic, we perform quantization of coeffs, input/output and this quantization is a form of noise

Also ideal SRRc has an infinite length; will need to look at different approaches to reduce the stopband attenuation to be greater than 40 dB

MER = modulation Error Ratio, this requirement is meant to ensure that PS and matched filter can work with each other to help us determine which signals we recv and sent

we take original 16 QAM symbols from input and compare them to the 16 QAM symbols from the outpyt

in our recv constellation, we can see that the output points will scatter around the ideal points in our input constellation

noise from quantization and channel; also extent of mismatch between PS and matched filter which generates a phenomenon known as ISI

if we design our PS and matched filter perfectly we will not have ISI; need to see how well the coeffs in match synergize with PS filter

MER is a measure of how spread out the recv points are from the original constellation points. If the points overlap too much, then it makes our system more error prone

Aim for the best MER where the points are cluster around

MER = avg(ideal received points/error in received points); convert this value into dB

create matlab script to tell you if your coefficients meet this spec