

## Digital Data transmission (Chapter 6+7)

9B-1

### Background

#### Sampling theorem:

A signal with bandwidth  $B$   
must be sampled at  $2B$ .

If we follow this rule, we can reconstruct it  
exactly.

$2B$  is the "Nyquist rate" (frequency)

$T_s = \frac{1}{2B}$  is the "Nyquist interval" (time)

### Signal reconstruction (6.1.1)

Given sampler —

send it through a low pass filter  
to reconstruct the baseband.

Text eq. 6.7 :

$$H(\omega) = T_s \operatorname{rect}\left(\frac{\omega}{4\pi B}\right)$$

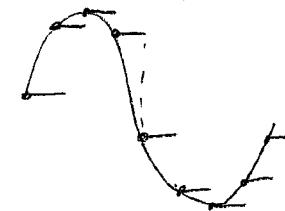
is the math description of an ideal  
low pass with bandwidth  $B$ , gain  $T_s$ .

Real filters only approximate this.

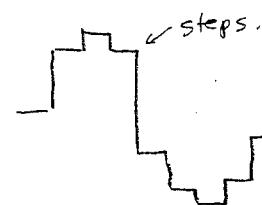
HW: 6.1 - 1, 2, 4

"Zero order hold" —

is a means of sampling



and reconstruction



"First order hold" — connects the dots with straight lines.



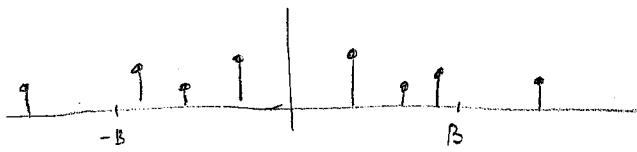
In practice — they use a zero order,  
followed by a low pass filter.

Practical difficulties ... (6.1.2)

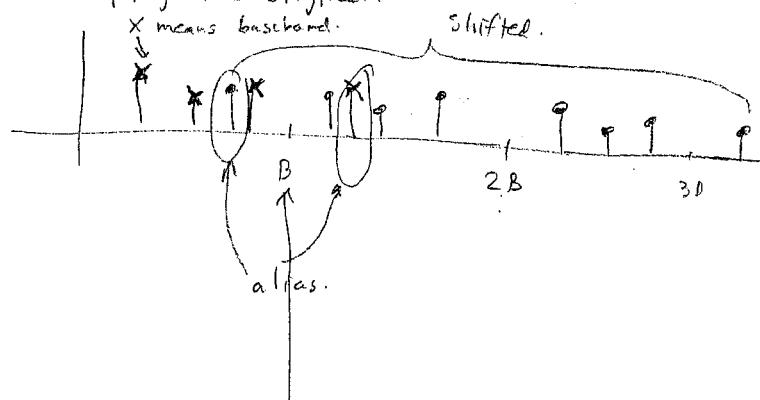
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You need a good filter to prevent aliasing.

Baseband spectrum:



Sample at  $2B$  → Sampling is frequency shifting, keeping the original.

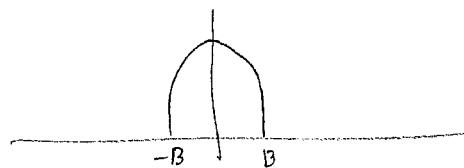


Folding frequency.

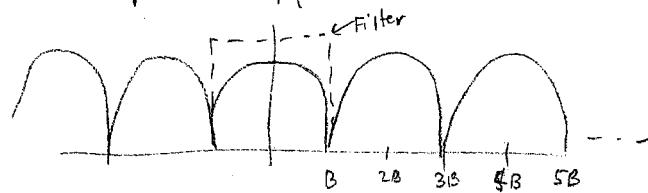
it "folds" around this.

Sampled spectrum is periodic,

Input spectrum

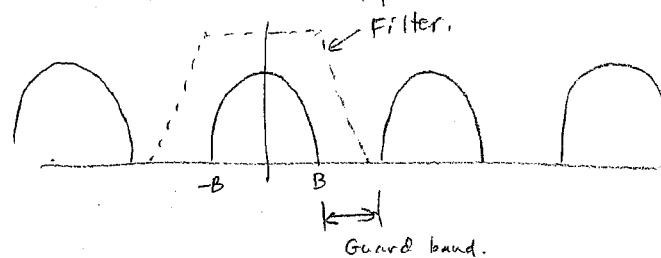


Sample at Nyquist rate ( $2B$ )



← Really need an ideal filter

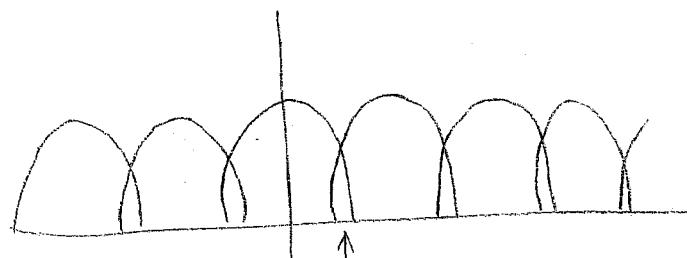
Sample above Nyquist rate



← Can use a less-than-ideal filter

← periodic in frequency — if impulse sampling.

Sample below Nyquist rate



Overlap → leads to aliasing.

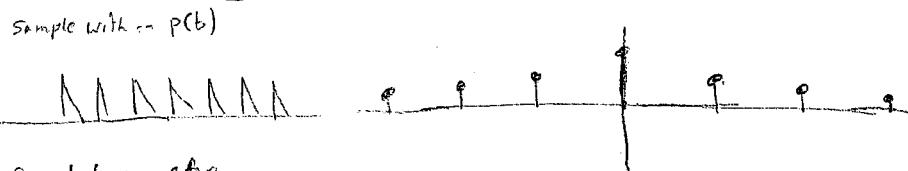
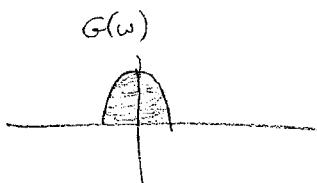
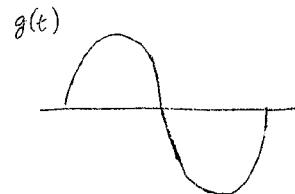
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## Practical Sampling

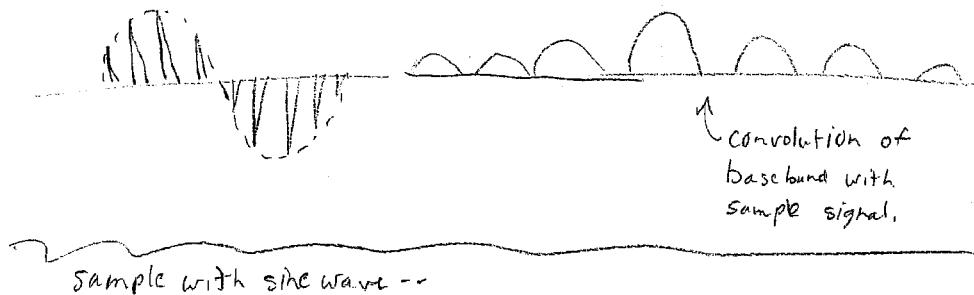
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If the sample waveform isn't a perfect impulse,  
the system still works -

The spectrum changes a little---



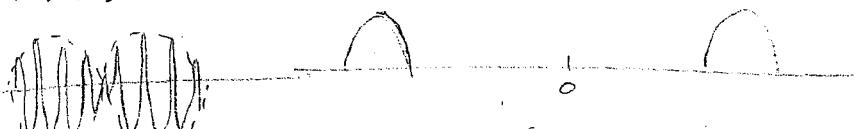
Sampled waveform



Sample with sine wave --



produces :



(does this look familiar?  
DSB-SC? )

## Maximum information rate

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Book - 6.1.3 says

"a maximum of  $2B$  independent pieces of information can be transmitted, error free, over a noiseless channel of bandwidth  $B$  Hz"

"A piece of information" here is a number,

not a "bit" which is used in information theory.

Justification ---

Sampling at the Nyquist rate produces  
2 samples per cycle.

Actually ---

Information is more properly measured in bits.

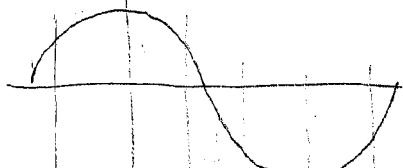
The amount of information that can be transmitted is limited by bandwidth and signal to noise ratio.

There are many coding schemes.

## Some "analog" coding schemes (6.1.4)

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The signal



Pulse - Amplitude Modulation

← The direct output of sampling, possibly by a switching type modulator.

Just filter to recover the baseband

Pulse width Modulation.

Vary the pulse width.  
Keep amplitude constant.

Just filter to recover the baseband, (but with a DC offset).

Pulse position modulation

Vary the position of fixed width pulses,  
Like FM,

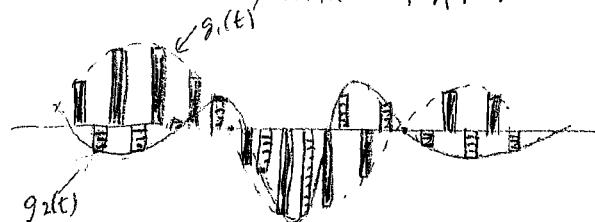
(applications of sampling theorem.)

## Time division multiplexing. (TDM)

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A way to send multiple signals on a channel--

consider 2 with PAM.



█ = Samples of  $g_1(t)$

▀ = Samples of  $g_2(t)$

Interleave the samples-

## Frequency division multiplexing (FDM)

The dual of TDM,

We already did this --

A bunch of SSB-SC or DSB-SC signals at different frequencies.

## Real digital methods

### Coding methods

#### 6.2 - Pulse code modulation

After sampling, generate a binary code.  
Transmit that.

#### 6.3 - Differential pulse code modulation.

Somehow, try to predict the next sample,  
Find the difference between the predicted  
and actual sample.

Generate code and transmit it.

#### 6.4 Delta modulation

A one bit version of DPCM,  
just say whether it is up or down.  
Need to oversample, but still usually  
less bandwidth used than DPCM.

## Transmission methods (Chapter 7)

Now that we have a code,  
how to transmit it?

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## Why digital?

(9 reasons on p. 263-264)

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1- Noise immunity. — (a) All we need to recover is  
1's and 0's. — If the level is in error,  
adjust it to match the allowed ones.

(b) Can use an error correcting code.

2- Can use repeaters.

With analog -- Every stage adds a little  
noise and distortion.

Digital — Repeaters can eliminate noise. —  
clean it up and retransmit.

3- Can use microprocessors, etc.

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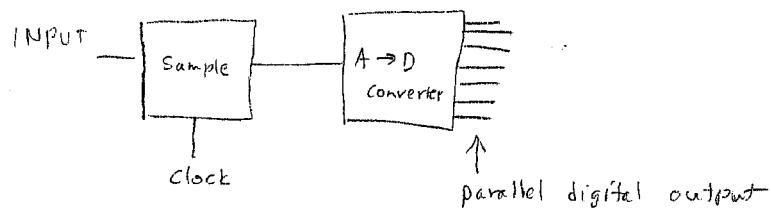
8- Reproduction can be exact.

No quality loss, if it is good enough.

## Pulse code modulation

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Introduction.

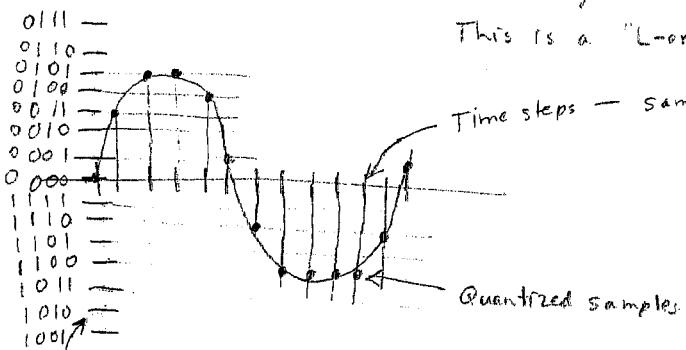


The input signal is sampled —

then quantized to L levels

and a number is generated for each sample

This is a "L-ary" digital signal.



Allowed quantization levels.

The codes for each level.

This one has 15 levels  
so it is a 15-ary digital signal

It uses a "2's complement" code.

(Example in book, p. 262, is a

"natural binary" code -- 0 to L-1)

These are not the only possible codes.

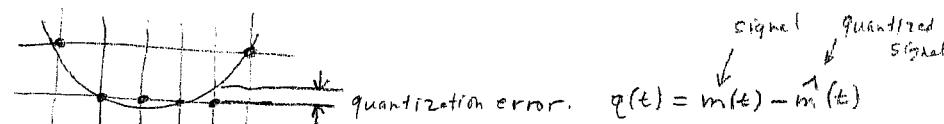
This signal is:

0000  
0011  
0101  
0101  
0100  
0001  
1110  
1100  
1100  
1100  
1110

Quantization -- only certain discrete levels are allowed.

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Quantization error is the error introduced because there are only discrete levels.



Quantization noise is the "signal" (sort of) produced by quantization error.



It isn't really noise, but we can make believe it is —

and calculate noise power, power spectral density, etc.

and therefore a signal to noise ratio.

$$\text{power} \rightarrow \frac{S}{N} = 3L^2 \frac{\text{mean square value}}{\text{peak value squared}}$$

For CD  $\approx 100$  dB for  $m_p^2 = m_p^2$   
(below peak signal)

Sine wave, max level --

$$\frac{m_p^2(t)}{m_p^2} = .5 \Rightarrow 97 \text{ dB.}$$