ELE 510 - COMMUNICATION THEORY

INSTRUCTOR: STEVEN KAY

TEXTBOOK: FUNDAMENTALS OF COMMUNICATION SYSTEMS - PROAKIS AND SALEHI

REFERENCES:

1) HAYKIN, COMMUNICATION SYSTEMS
2) SHANMUGAM, DIGITAL AND ANALOG COMMUNICATION SYSTEMS
3) PROAKIS, DIGITAL COMM.
4) WOZENCRAFT, JACOBS, PRINCIPLES OF COMM. ENGINEERING

COURSE OUTLINE: SEE TABLE OF CONTENTS

CHAPTER 1 - INTRODUCTION
CHAPTER 7 - ANALOG-TO-DIGITAL CONVERSION
CHAPTER 8 - DIGITAL MODULATION
CHAPTER 9 - DIGITAL TRANSMISSION
CHAPTER 10 - TRANSMISSION VIA CARRIER
CHAPTER 12 - INFORMATION THEORY
CHAPTER 11 - SELECTED TOPICS (MULTIPATH, OFDM, SPREAD SPECTRUM)

WILL COVER SECTIONS (NOT ALL) OF THESE CHAPTERS

NOTES: WILL BE POSTED ON-LINE.
PLEASE READ BEFORE CLASS!

PREREQS: ELE 509 OR EQUIVALENT
(SEE CHAPTER 5 FOR REVIEW)
WILL ALSO NEED SIGNALS AND
SYSTEMS (SEE CHAPTER 2
FOR REVIEW)

GRADING: NO TESTS ☺

1) HOMEWORK ASSIGNED WEEKLY - DUE
FOLLOWING WEEK - GRADES AS
"✓" (OK) OR "✓-" (NOT OK).
REASONABLE EFFORT REQUIRED OR
GRADE LOWERED!

2) SMALL DESIGN PROJECT
(DATA GIVEN - MUST DECODE) - 20%

3) MAJOR PROJECT - 80%

a) LITERATURE SURVEY
b) ANALYSIS OF COMM. SYSTEM,
TECHNIQUE, ETC.
c) COMPUTER SIMULATION
d) OR OTHERS?

PROPOSAL (ONE PAGE) REQUIRED
FORMAL REPORT + ORAL PRESENTATION
DUE LAST DAY OF CLASS
(SEE ME FOR IDEAS)

OFFICE HOURS: TUES, FRI 2-4 PM
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INTRODUCTION

READ CHAPTER 1

WHY STUDY COMMUNICATIONS?

VOICE - TELEPHONES, CELL PHONES
DATA - COMPUTER FILES, FAX, EZ PASS
INTERNET, IPODS, CDS, DVDS
VIDEO - MOVIES, TV, INTERNET,
SURVEILLANCE CAMERAS

ETC, ETC, ETC

BASIC ELEMENTS - INFORMATION,
TRANSMISSION, RECEPTION

INFORMATION - VOICE SIGNAL
TRANSMISSION - TRANSDUCER + CHANNEL
RECEPTION - OUTPUT TRANSDUCER

NOTE: INFORMATION (VOICE, DATA, VIDEO) IS ALWAYS UNKNOWN TO RECEIVER ⇒ MODELED AS RANDOM PROCESS

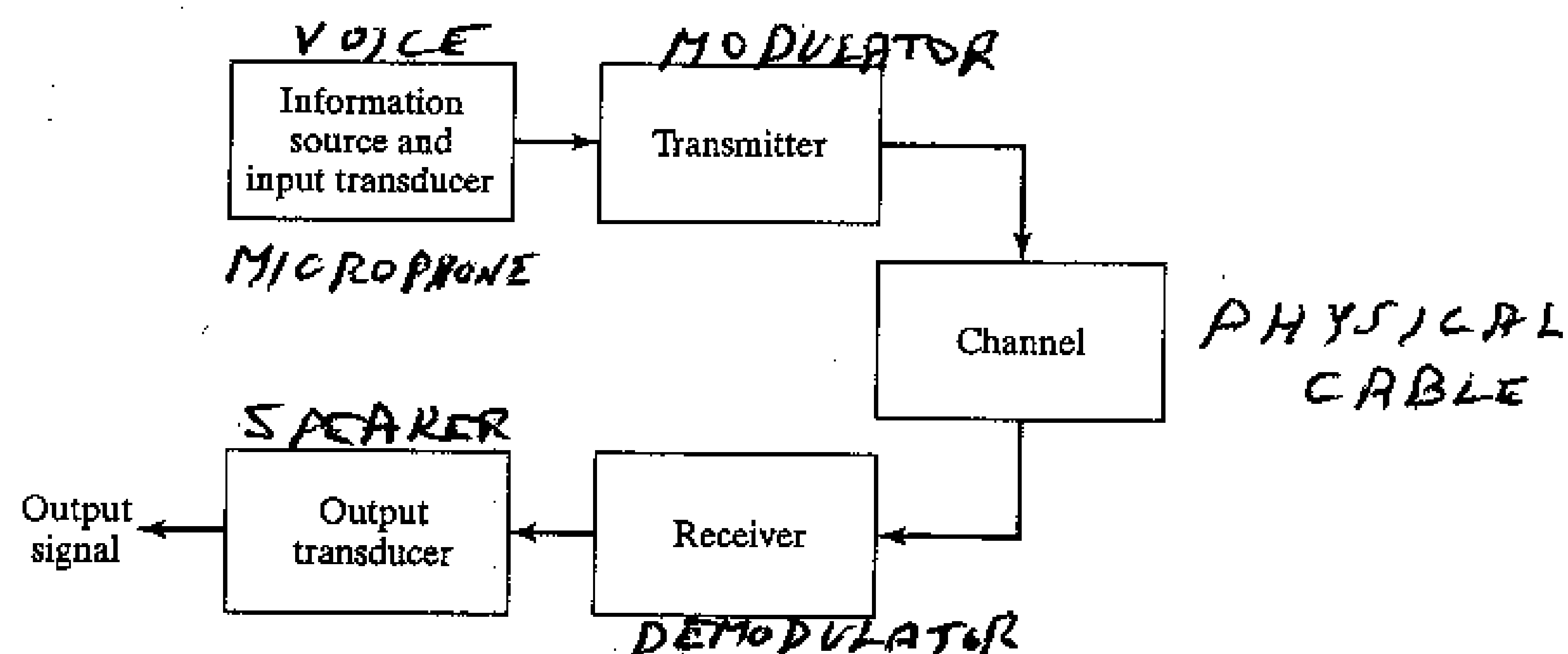


Figure 1.1 Functional diagram of a communication system.

CHANNEL COULD BE:
WIRE, CABLE, FREE SPACE (RADIO), OPTICAL FIBER, ETC.

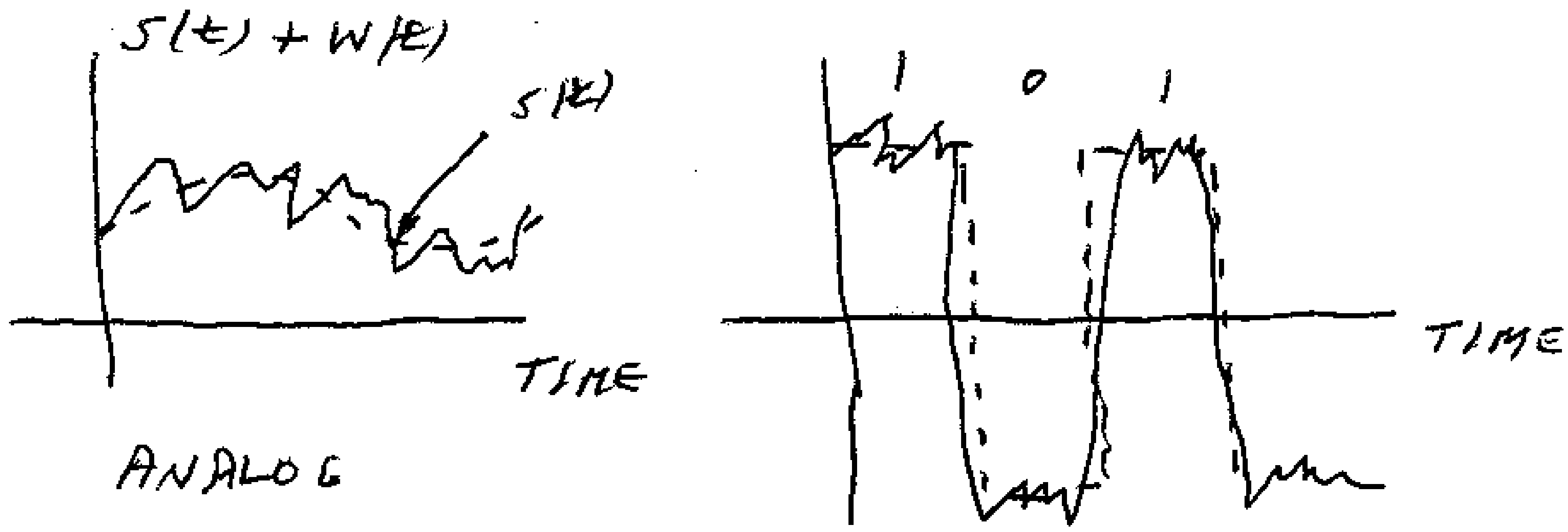
ANALOG VS. DIGITAL COMM. -

ANALOG-MUSIC - $\underbrace{\text{AM, FM}}_{\text{ANALOG (CHAPTERS 3, 4)}}$, TV ← CONVERTS TO DIGITAL SOON

DIGITAL-MUSIC - CDs, IPOD

WHY DIGITAL ?

1) CD VS PHONOGRAPH RECORD - DIGITAL PROVIDES VIRTUALLY ERROR FREE STORAGE AND TRANSMISSION -



ANALOG

DIGITAL

WHICH SIGNAL IS EASIER TO RECOVER ?

IN DIGITAL WE USE REGENERATIVE REPEATERS, IN ANALOG WE USE ANALOG AMPLIFIERS.

REPEATERS - GETS RID OF NOISE

AMPLIFIERS - ADDS NOISE

2) IN DIGITAL FORM CAN EASILY MANIPULATE SIGNAL - STORAGE, ENCRYPTION, REDUNDANCY, ETC.

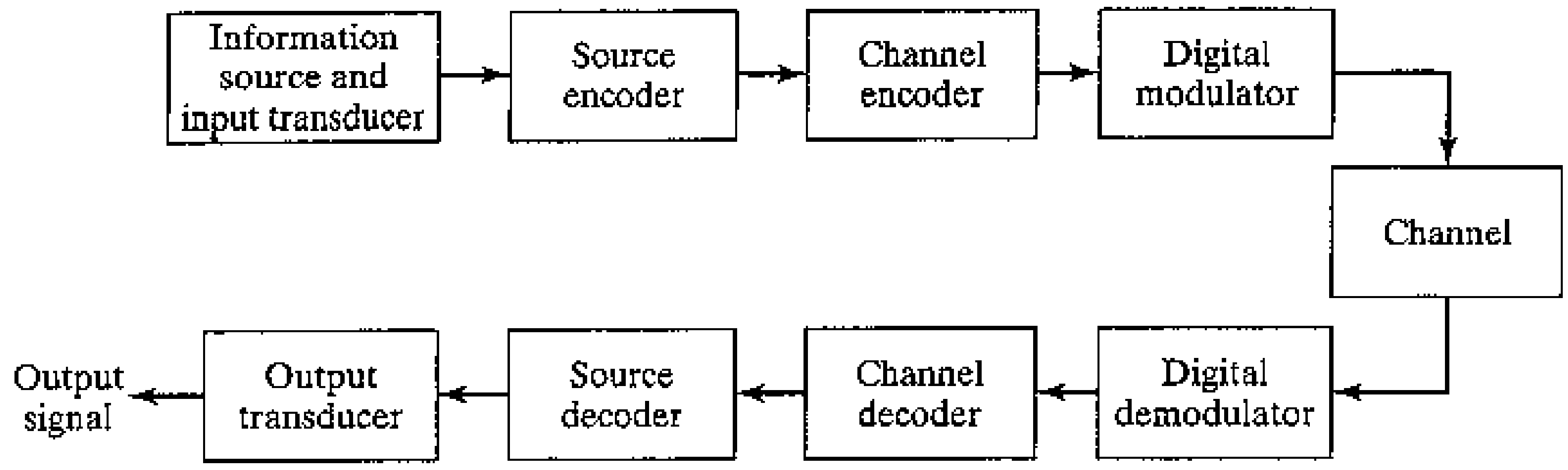
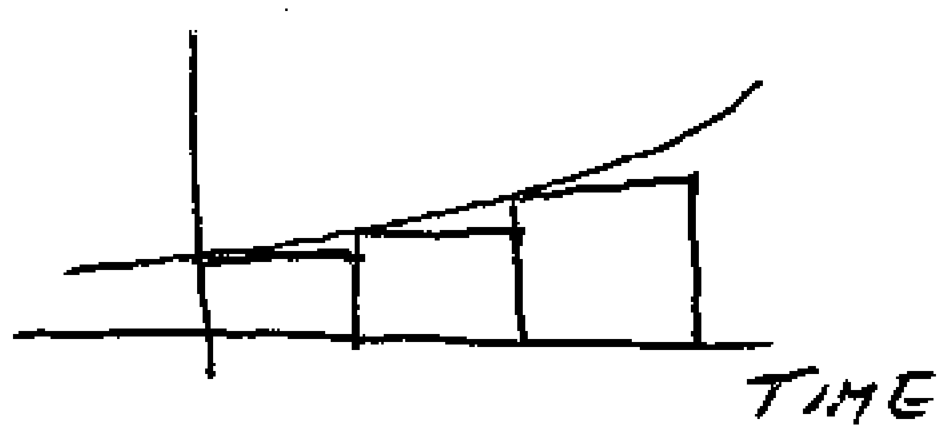


Figure 1.2 Basic elements of a digital communication system.

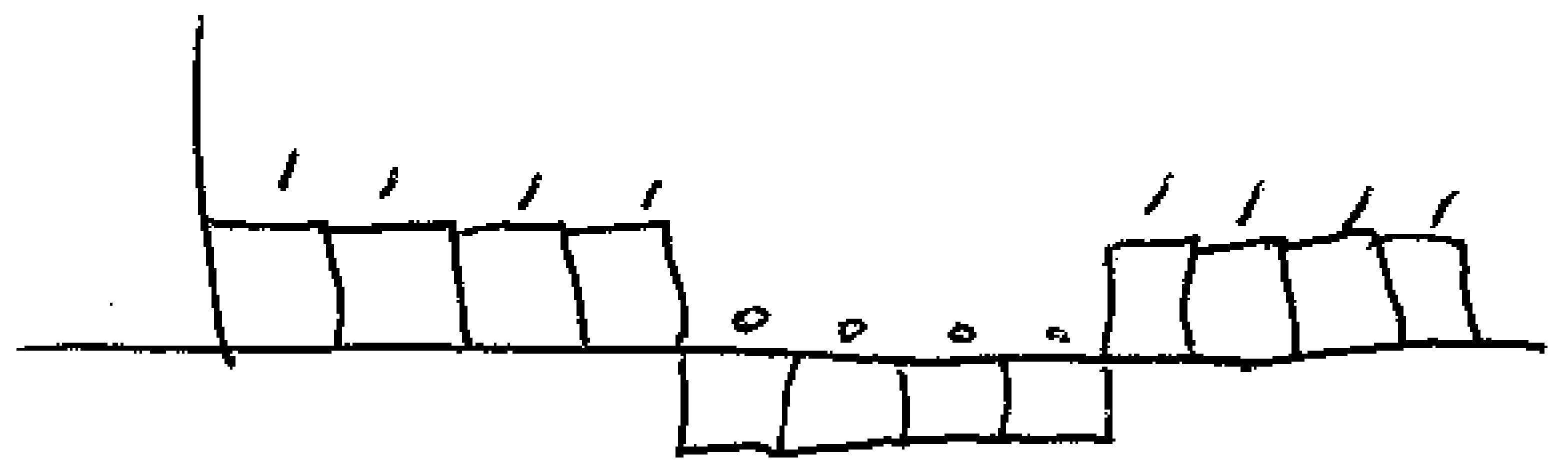
OUR FOCUS

SOURCE ENCODER - REMOVE REDUNDANCY AND CONVERT TO BIT STREAM



ANALOG SOURCE

CONVERT TO PULSES



DIGITAL SOURCE

CONVERT TO 101

⇒ DATA COMPRESSION (NEED LESS BANDWIDTH TO TRANSMIT)

CHANNEL ENCODER - ADD REDUNDANCY TO COMBAT NOISE

EXAMPLE : SOURCE TRANSMITS ... 101 ...

ENCODER CONVERTS TO

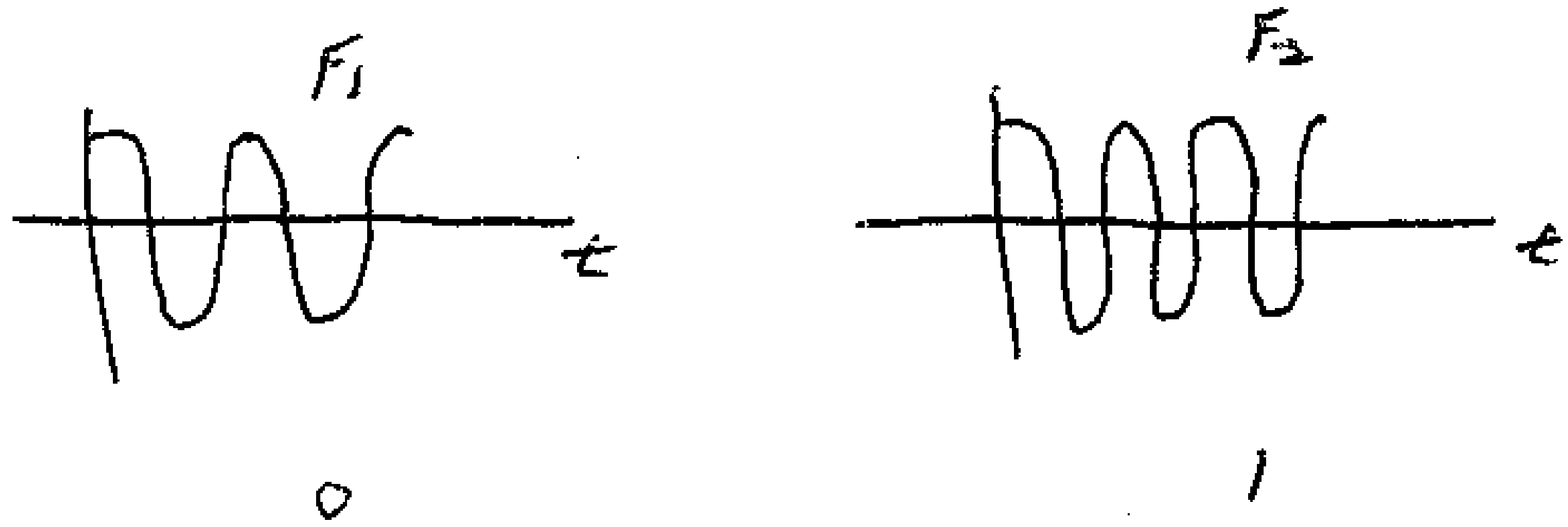
... 111 000 111 ...

DECODE AS 1 IF MAJORITY OF BITS ARE 1'S AND AS 0 OTHERWISE

NOTE THAT BIT RATE DECREASES BY 3 BUT ERROR RATE ALSO DECREASES.

BIT RATE / ERROR RATE TRADEOFF. - WILL SEE THIS WITH ALL COMM. SYSTEMS

DIGITAL MODULATOR - CONVERTS BITS TO WAVEFORMS TO ALLOW TRANSMISSION THROUGH CHANNEL



FREQUENCY SHIFT KEYING

FOR RADIO TRANSMISSION NEED ANTENNA LENGTH = $L = \lambda/4$

$c = f_0 \lambda$
↑ SPEED OF LIGHT

$$\Rightarrow \quad f_0 = c/\lambda = \frac{c}{4L}$$

IF $L = 3''$ (CELL PHONE),

$$f_0 = \frac{3\times10^8}{4(3/39.37)} = 0.984 \text{ GHz}$$

10^9

NOTE: IF WE USED $M = 32$ FREQUENCIES, WE COULD REPRESENT $\log_2 M = 5$ BITS

$\longrightarrow t$

$\ldots\underbrace{01011}_{11}\,\underbrace{10111}_{23}\ldots \longrightarrow \boxed{\text{DIGITAL MOD}} \longrightarrow$ $\cos(2\pi f_{11} t)$ $\quad t_1 \le t \le t_1 + T$
FOLLOWED BY
$\cos(2\pi f_{23} t)$

IF SOURCE PRODUCES BITS AT RATE $R = 1/T_b$ BITS/SEC, THEN OUTPUT OF MODULATOR CAN BE

$$\cos(2\pi f_{11} t) \qquad t_1 \le t \le t_1 + 5T_b$$

WAVEFORM IS 5 TIMES LONGER

IN GENERAL, FOR k BITS NEED $M = 2^k$ WAVEFORMS.

<u>DIGITAL DEMODULATOR</u> - CONVERTS RECEIVED WAVEFORM BACK TO 0 OR 1 (BINARY) OR GROUP OF 5 BITS (M-ARY, $M = 32$), HOPEFULLY WITHOUT ERRORS

<u>CHANNEL DECODER</u> - REMOVES REDUNDANCY BITS

<u>SOURCE DECODER</u> - UNDOES EFFECT OF SOURCE ENCODER, FOR EXAMPLE, FOR TEXT IF WE HAD TRANSMITTED

"QUICK" $\xrightarrow{\text{ENCODER}}$ QICK $\xrightarrow{\text{DECODER}}$ QUICK

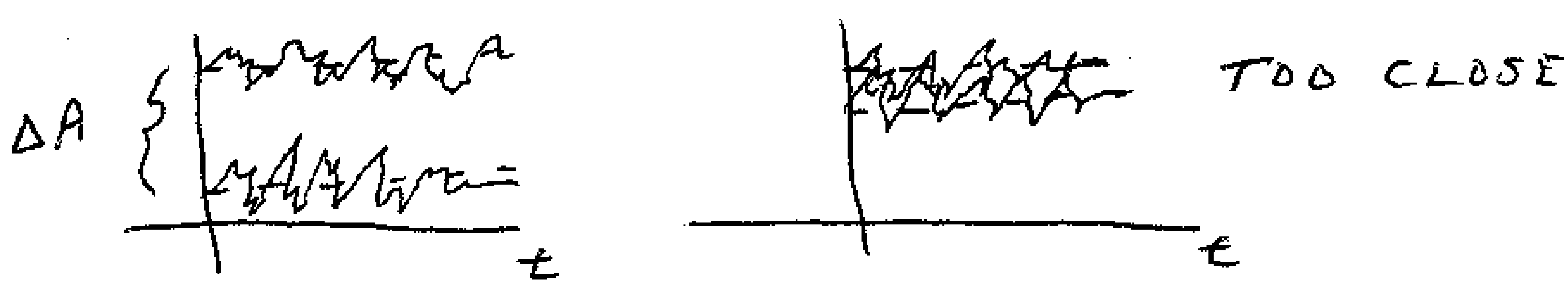
SOME HISTORY

<u>NYQUIST</u> (1924) - HOW FAST CAN WE TRANSMIT WAVEFORMS THROUGH BANDLIMITED CHANNEL? ANSWER - FOR W Hz CHANNEL CAN TRANSMIT AT 2W WAVEFORMS / SEC USING

$$g(t) = \frac{\sin 2\pi W t}{2\pi W t}$$

(RELATED TO SAMPLING THEOREM).
DIDN'T CONSIDER <u>NOISE</u>.

<u>HARTLEY</u> (1928) - FOR W Hz CHANNEL AND POWER CONSTRAINT CAN ONLY RELIABLY DISCERN AMPLITUDES IF FAR ENOUGH APART DUE TO NOISE



ΔA DEPENDS ON NOISE LEVEL.

SHANNON (1948) - CONSIDERED MATHEMATICALLY HOW MUCH INFORMATION COULD BE TRANSMITTED OVER CHANNEL WITH BANDWIDTH W Hz, POWER OF TRANSMITTER P WATTS, AND NOISE LEVEL $\frac{N_0}{2}$ WATTS/Hz

$$C = W \log_2 \left(1 + \underbrace{\frac{P}{W N_0}}_{SNR}\right)$$

CHANNEL CAPACITY

IF $R < C \Rightarrow$ ERRORLESS WITH $P_e \to 0$
IF $R > C \Rightarrow$ ERRORS WITH $P_e \to 1$

SHANNON'S WORK AT BELL LABS WAS FUNDAMENTAL AND GROUNDBREAKING - WILL STUDY LATER IN CHAPTER 12 (INFORMATION THEORY)

INTERESTINGLY (FRUSTRATINGLY!) HIS THEOREMS WERE EXISTENCE ONLY - NOT CONSTRUCTIVE.

TODAY HIS THEOREMS AND FUNDAMENTAL LIMITS HAVE BEEN ACHIEVED!

SEE WIKIPEDIA FOR MORE INFO.

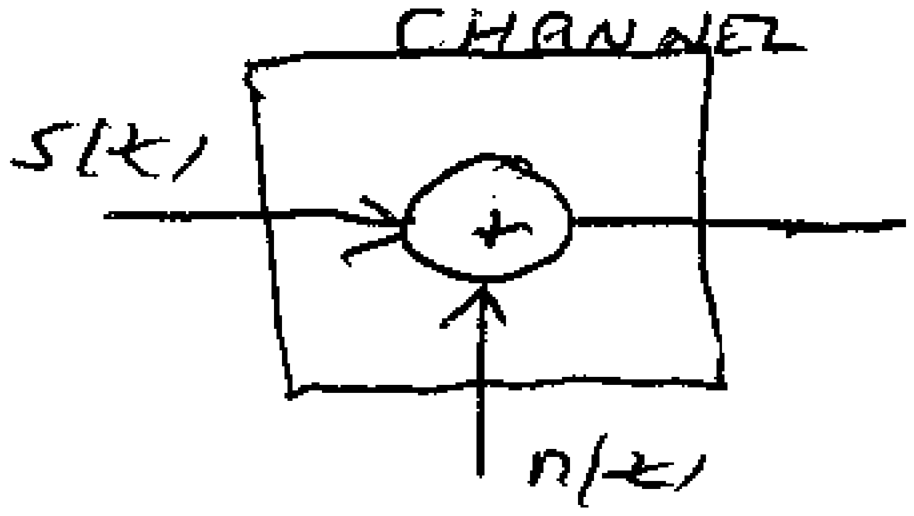
READ SECTION 1.3 - CHARACTERISTICS OF COMMUNICATION CHANNELS
WIRELINE, FIBER OPTICS, WIRELESS EM, UNDERWATER, STORAGE

CDS, DVDS, MAGNETIC TAPE/STRIPS, COMPUTER MEMORY ARE ALL STORAGE CHANNELS - ALL OUR DISCUSSIONS APPLY TO THEM AS WELL (DIFFERENCE IS REAL-TIME VS. NON REAL-TIME COMM. SYSTEM)

COMM. SYSTEM MODELS

TWO BASIC MODELS - LARGE BANDWIDTH OR NOT

1) LARGE BANDWIDTH - NO WAVEFORM DISTORTION IN ABSENCE OF NOISE



$r(t) = s(t) + n(t)$

NOISE

WILL ASSUME $n(t)$ IS WHITE GAUSSIAN NOISE (CALLED ADDITIVE WGN OR AWGN CHANNEL) - EXAMPLE IS SPACE CHANNEL

2) NOT LARGE BANDWIDTH - WAVEFORM DISTORTION OCCURS

$$r(t) = \int_{-\infty}^{\infty} h(\tau)\, s(t-\tau)\, d\tau + n(t)$$

LINEAR
TIME INVARIANT
FILTER (SYSTEM)

EXAMPLE IS WIRELINE CHANNEL.

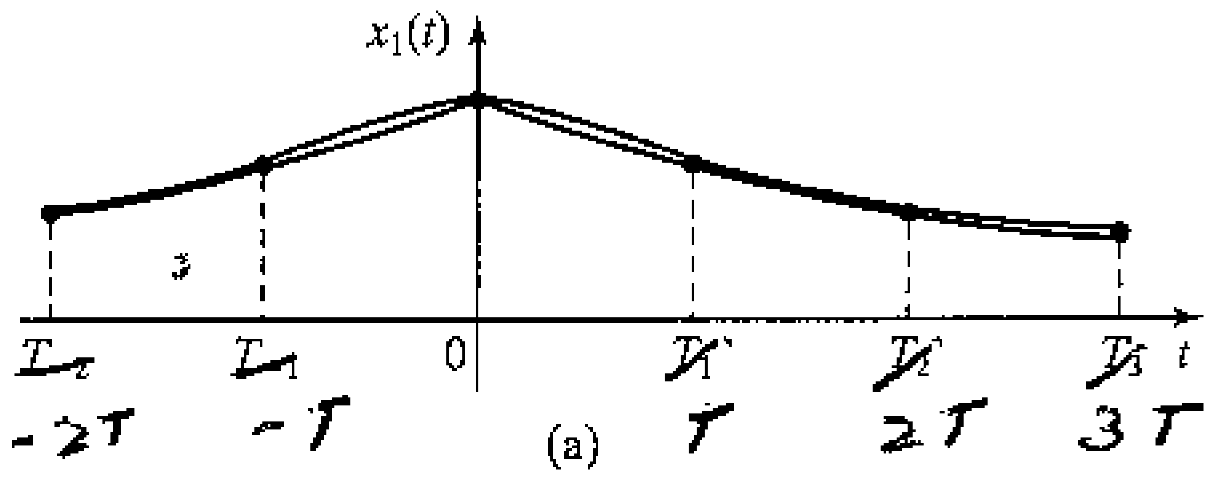
CHAPTER 7 - ANALOG-TO-DIGITAL CONVERSION (7.1 - 7.5)*

CONSIDER SPEECH TRANSMISSION -
SPEECH IS INFORMATION, MICROPHONE IS INPUT TRANSDUCER, SOURCE ENCODER CONVERTS ANALOG WAVEFORM TO DIGITAL REPRESENTATION

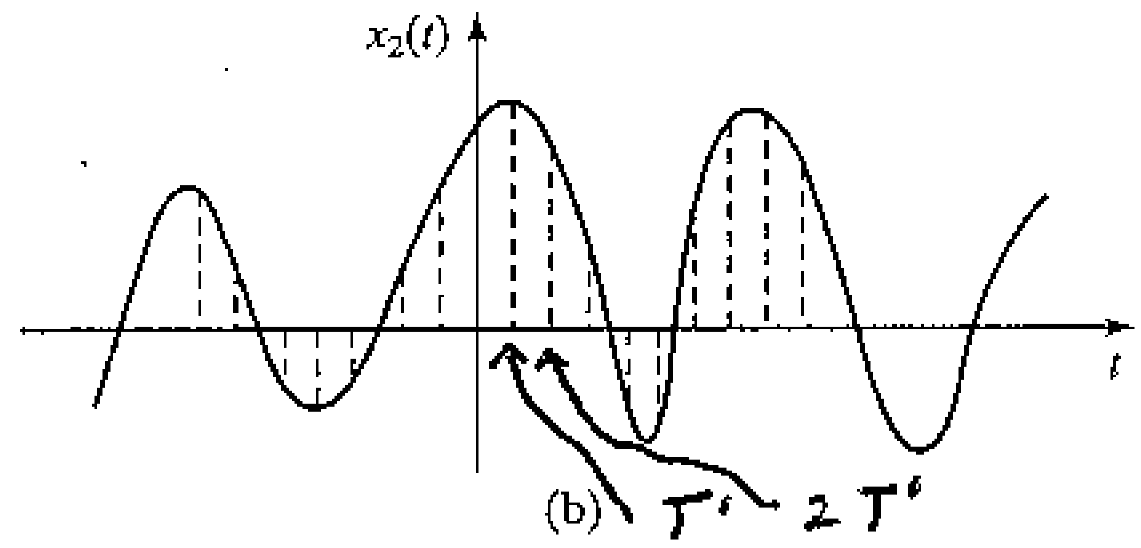
STEPS :
1) SAMPLING
2) QUANTIZING
3) ENCODING

CONSIDER SAMPLING FIRST.

* SELECTED PARTS



CHANGES SLOWLY



CHANGES RAPIDLY

$T' < T$

Figure 7.1 Sampling of signals.

RATE OF CHANGE DEPENDS ON BANDWIDTH OF SIGNAL (FOURIER TRANSFORM MEASURES BANDWIDTH)

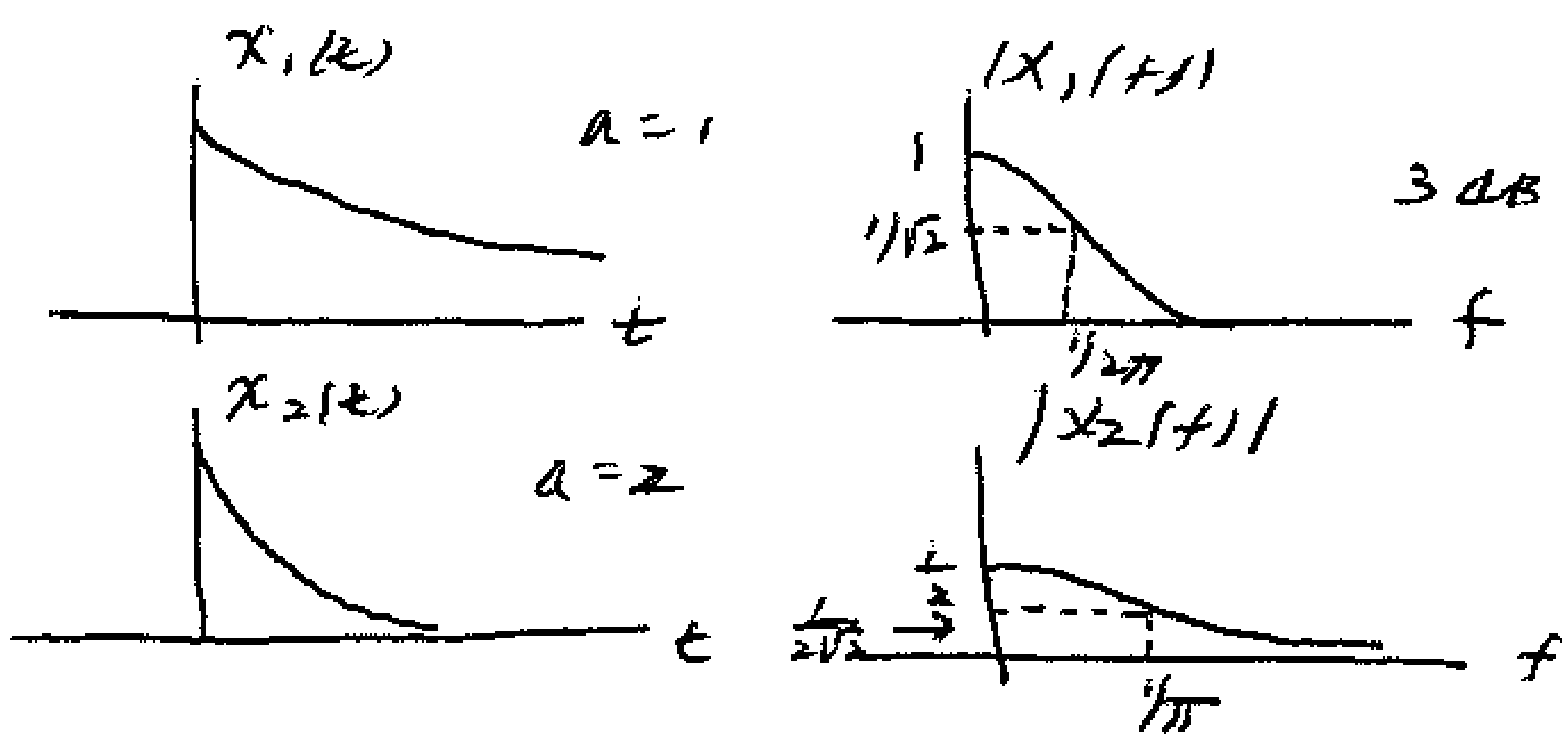
EXAMPLE: $x(t) = e^{-at}$ $t \geq 0$ $a > 0$

$\phantom{x(t) = {}} 0$ $t < 0$

$$X(f) = \int_0^\infty e^{-at} e^{-j2\pi f t} \, dt$$

$$= \left. \frac{e^{-(a+j2\pi f)t}}{-(a+j2\pi f)} \right|_0^\infty = \frac{1}{a+j2\pi f}$$

$$|X(f)| = \frac{1}{\sqrt{a^2+(2\pi f)^2}} = \frac{1/a}{\sqrt{1+(2\pi f/a)^2}}$$



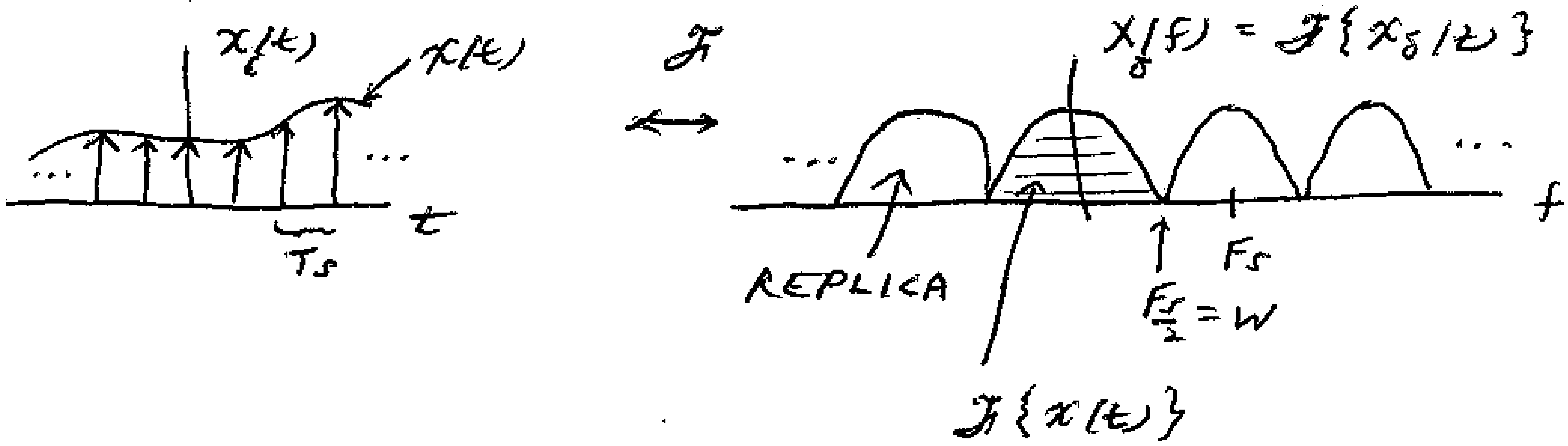
3 dB BANDWIDTH $= \frac{a}{2\pi}$

SAMPLING THEOREM - ASSUME $x(t)$ IS BANDLIMITED TO W Hz ($X(f) = 0$ FOR $|f| \geq W$), IF WE TAKE SAMPLES $\{\ldots, x(-Ts), x(0), x(Ts), \ldots\}$ THEN TO BE ABLE TO RECOVER $x(t)$ FROM THESE SAMPLES, WE REQUIRE

$$T_S \leq \frac{1}{2W}$$

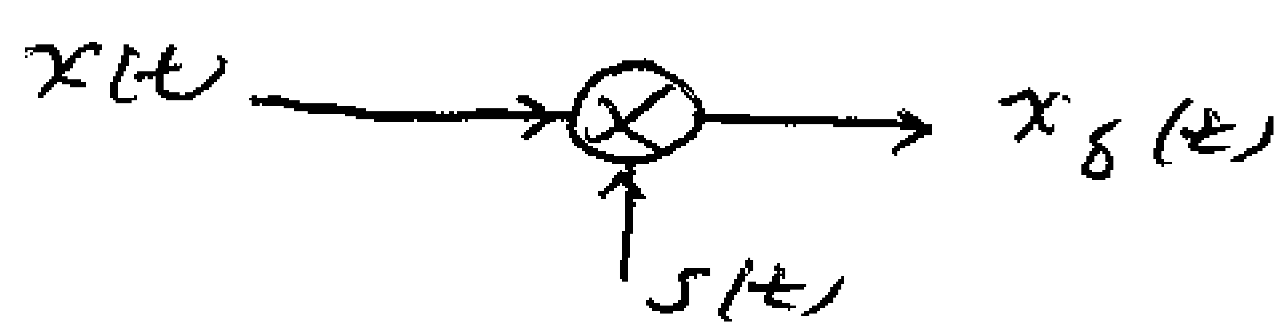
OR WE MUST SAMPLE AT A RATE OF $f_s = 1/T_s \geq 2W$ SAMPLES/SEC.

WHY? SAMPLING IN TIME CAUSES FOURIER TRANSFORM TO BE REPLICATED IN FREQUENCY



TO RECOVER $x(t)$ FROM $x_\delta(t)$ OR $X(f)$ FROM $X_\delta(f)$ (EQUIVALENTLY), JUST GET RID OF REPLICAS $\Rightarrow$ LOW PASS FILTER $x_\delta(t)$.

PROOF : SAMPLING OPERATION MODELED AS



WHERE $s(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$

1 = STRENGTH OF IMPULSE (NOT HEIGHT)

TO FIND FOURIER TRANSFORM OF $s(t)$, USE "ROUNDABOUT" APPROACH ($S(f)$ DOESN'T ACTUALLY EXIST MATHEMATICALLY)

USE FOURIER SERIES REPRESENTATION FOR PERIODIC SIGNAL OF PERIOD T_s:

$$s(t) = \sum_{n=-\infty}^{\infty} c_n e^{j2\pi n f_0 t} \qquad f_0 = 1/T_s$$

$$c_n = \frac{1}{T_s} \int_{-T_s/2}^{T_s/2} s(t) e^{-j2\pi n f_0 t}\, dt$$

HERE $s(t) = \delta(t) \quad |t| \le T_s/2$

$$\Rightarrow \quad c_n = 1/T_s \Rightarrow s(t) = \sum_{n=-\infty}^{\infty} \frac{1}{T_s} e^{j2\pi n f_0 t}$$

BUT $e^{j2\pi n f_0 t} \overset{\mathcal{F}}{\longleftrightarrow} \delta(f - nf_0)$

$$= \delta(f - n/T_s)$$

$$\Rightarrow \quad S(f) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} \delta(f - n/T_s)$$

NOW $x_\delta(t) = s(t)\, x(t)$

$$X_\delta(f) = S(f) * X(f)$$

CONVOLUTION IN f

$$X_1(f) * X_2(f) = \int_{-\infty}^{\infty} X_1(\zeta)\, X_2(f-\zeta)\, d\zeta$$

$$\Rightarrow \quad X_\delta(f) = \int_{-\infty}^{\infty} S(\zeta)\, X(f-\zeta)\, d\zeta$$

$$= \int_{-\infty}^{\infty} \frac{1}{T_s} \sum_{n=-\infty}^{\infty} \delta\left(\zeta - \frac{n}{T_s}\right) X(f-\zeta)\, d\zeta$$

$$= \frac{1}{T_s} \sum_{n=-\infty}^{\infty} \int_{-\infty}^{\infty} \delta\left(\zeta - \frac{n}{T_s}\right) X(f-\zeta)\, d\zeta$$

$$\therefore\ X_\delta(f) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} \underbrace{X\left(f - \frac{n}{T_s}\right)}$$

$$\ldots,\ X(f + 1/T_s),\ X(f),\ X(f - 1/T_s),\ \ldots$$

REPLICA $\qquad\qquad$ REPLICA

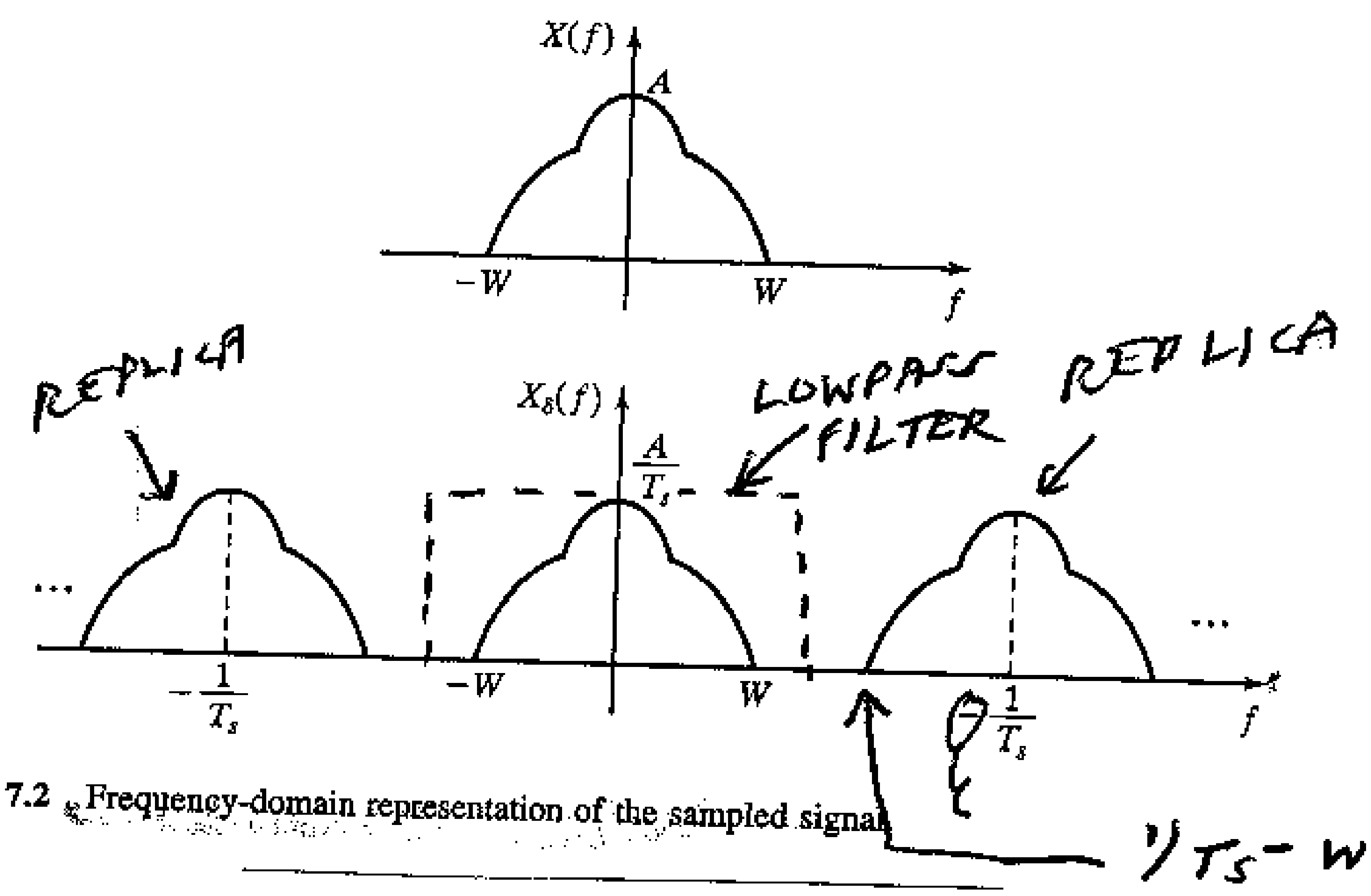


Figure 7.2 Frequency-domain representation of the sampled signal.

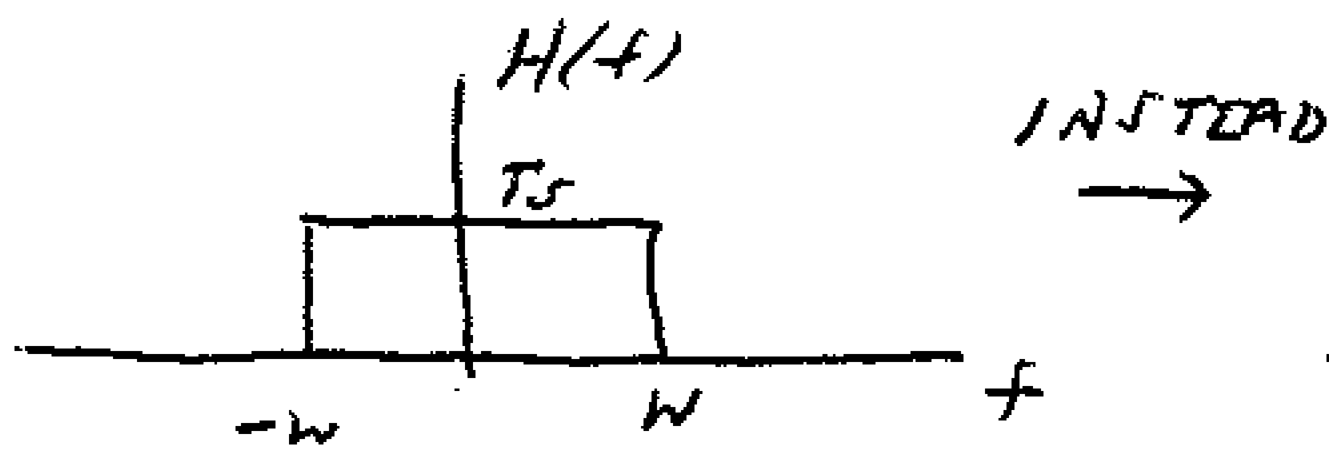
TO RECOVER $X(f)$ (AND HENCE $x(t)$) NEED TO "KNOCK OUT" REPLICAS $\Rightarrow$ USE LOWPASS FILTER. TO DO SO NEED

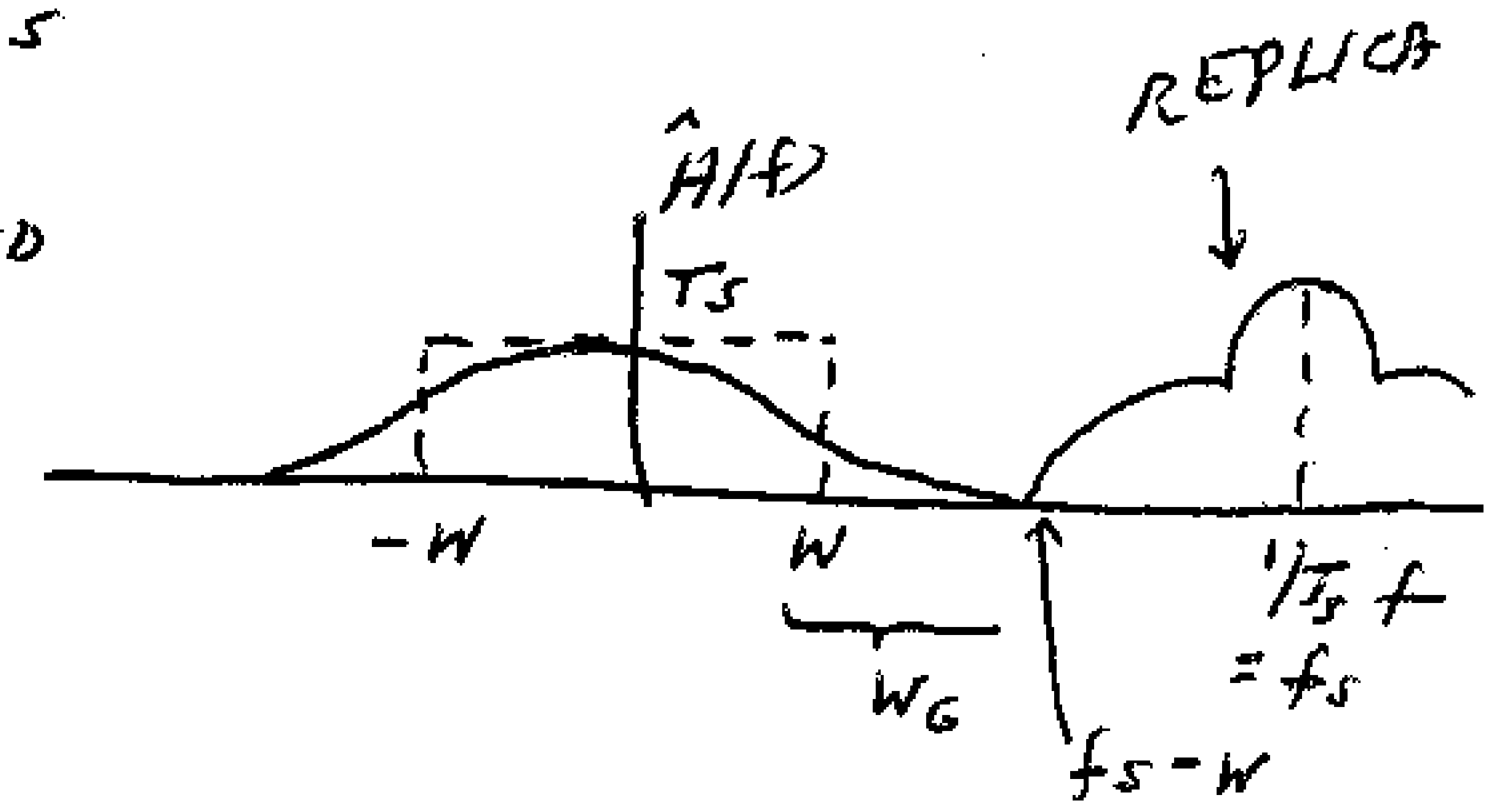
$1/T_s - W \geq W$ OR $1/T_s \geq 2W$ OR

$f_s \geq 2W$ SAMPLES/SEC

OTHERWISE REPLICAS OVERLAP AND WE CAN'T FILTER OUT.

ALTHOUGH THEORETICALLY POSSIBLE TO USE $f_s = 2W$, WOULD REQUIRE <u>PERFECT</u> LOWPASS





W_G = GUARD BAND

$\Rightarrow f_s = 2W + W_G$

$W + W_G = f_s - W$

TO RECOVER $x(t)$ USING PERFECT LOWPASS FILTER:

$$x(t) = \mathcal{F}^{-1}\{H(f)X_\delta(f)\}$$

$$= h(t) * x_\delta(t)$$

$$= h(t) * \sum_{n=-\infty}^{\infty} x(nT_s)\,\delta(t - nT_s)$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s)\underbrace{\left[h(t) * \delta(t - nT_s)\right]}_{h(t - nT_s)}$$

BUT $h(t) = \mathfrak{F}^{-1}\{H(f)\}$

$$= \int_{-W}^{W} T_s \, e^{j2\pi f t} \, df$$

$$= T_s \int_{-W}^{W} \cos 2\pi f t \, df \qquad \text{WHY?}$$

$$= T_s \left. \frac{\sin 2\pi f t}{2\pi t} \right|_{-W}^{W}$$

$$= T_s \left[\frac{\sin 2\pi W t}{2\pi t} - \frac{\sin(-2\pi W t)}{2\pi t} \right]$$

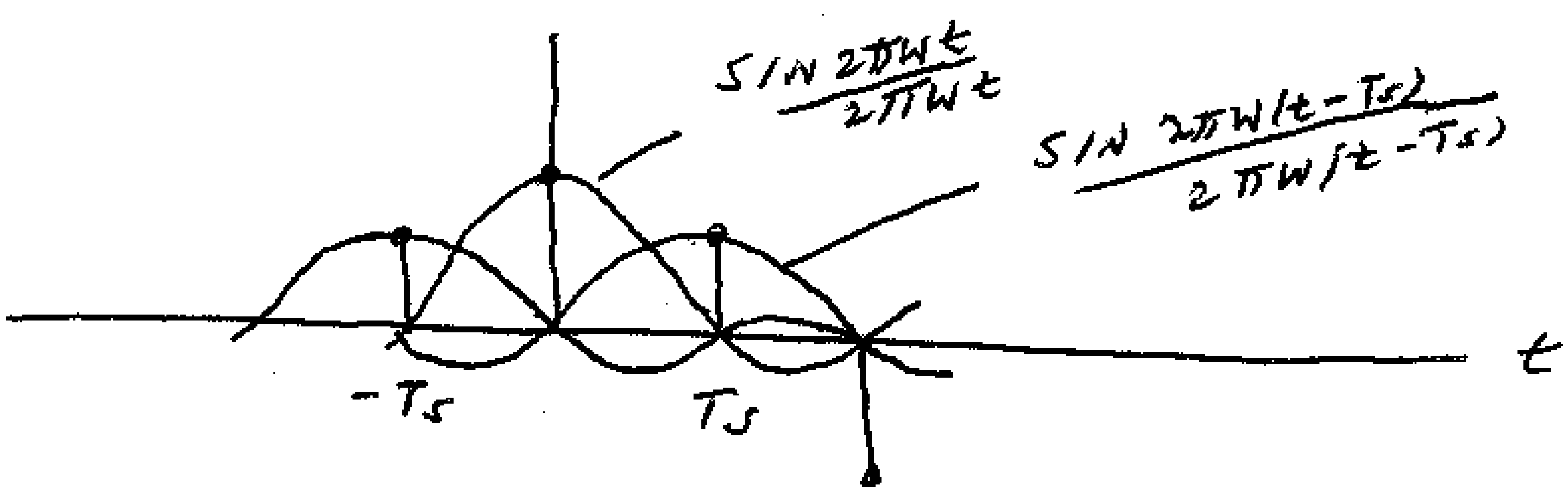
$$= 2 T_s \frac{\sin 2\pi W t}{2\pi t}$$

$$= \underbrace{2 W T_s}_{=1} \frac{\sin 2\pi W t}{2\pi W t}$$

$$= \frac{\sin 2\pi W t}{2\pi W t}$$

$$\therefore \quad x(t) = \sum_{n=-\infty}^{\infty} x(nT_s) \frac{\sin[2\pi W (t - nT_s)]}{2\pi W (t - nT_s)}$$

CALLED <u>INTERPOLATION</u> <u>FORMULA</u>



NOTE THAT AT SAMPLE TIMES $t = \pm T_s, \pm 2T_s, \ldots$

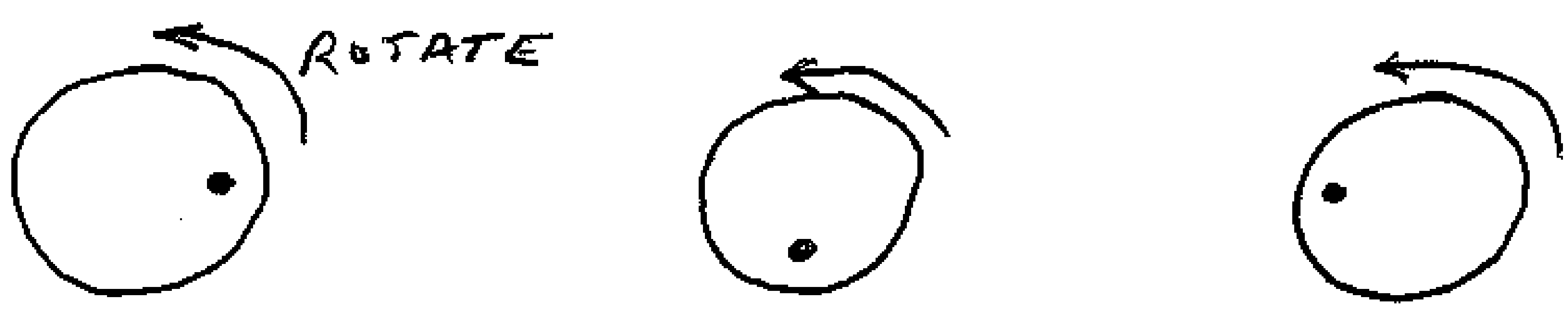
$$\frac{\sin 2\pi W t}{2\pi W t} = \frac{\sin 2\pi W n T_s}{2\pi W n T_s} = \frac{\sin n\pi}{n\pi} = 0$$

FOR $T_s = 1/2W$.

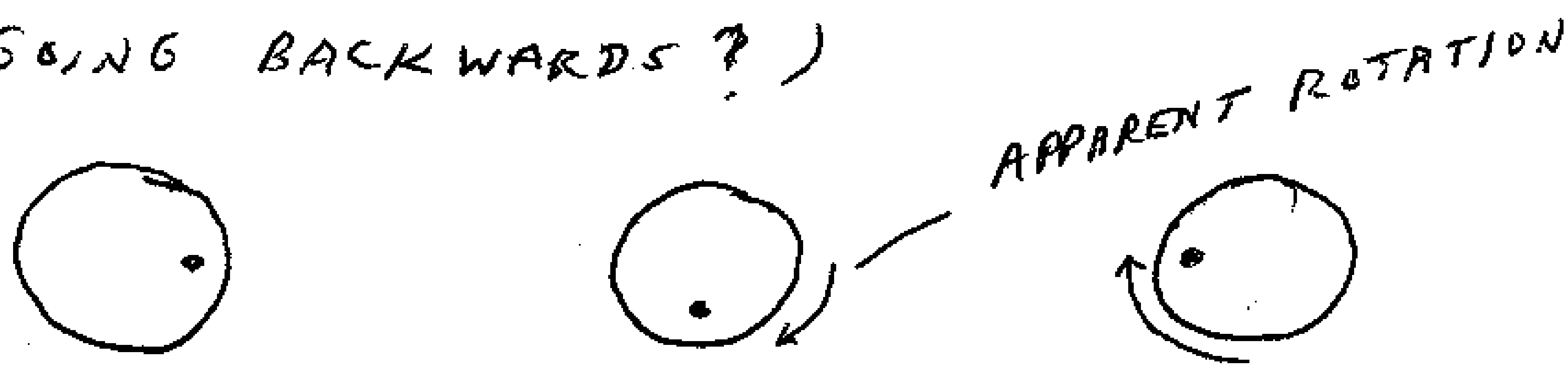
NOTE: IF $1/T_s < 2W$, WE HAVE OVERLAPPED SPECTRA - CALLED <u>ALIASING</u>

<u>EXAMPLE</u>: TELEPHONE SPEECH IS USUALLY FIRST LOWPASS FILTERED TO $W = 3400$ Hz $\Rightarrow$ NEED $f_s \geq 2W = 6800$. TYPICALLY $W_G = 1200$ Hz. THUS, $f_s = 2W + W_G = 8000$ SAMPLES/SEC.

ASIDE: TOYOTA WHEELS IN COMMERCIAL



ROTATE 3/4 TURN PER FRAME (WHEEL GOING BACKWARDS?)

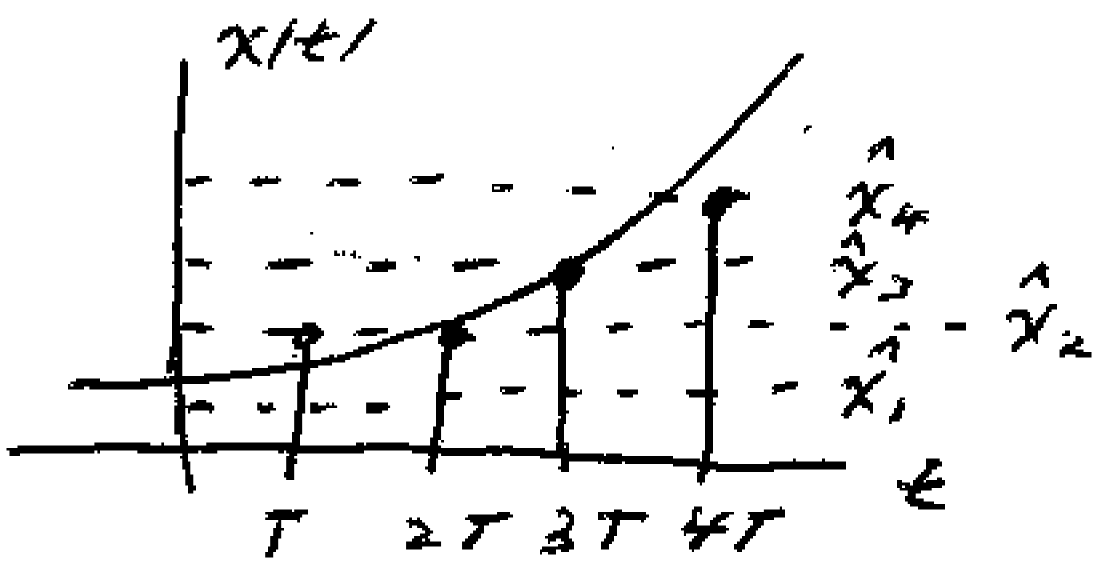


HOW MANY FRAMES PER ONE ROTATION DO YOU NEED TO AVOID "VISUAL ALIASING"?

QUANTIZATION

CANNOT TRANSMIT REAL NUMBERS $-\infty < x < \infty$ SINCE WE WOULD NEED ∞ NUMBER OF BITS.

COULD, HOWEVER, TRANSMIT ONE OF N NUMBERS



$N = 4$

$\hat{x}_i$ ARE QUANTIZED LEVELS

EACH QUANTIZED LEVEL IS ENCODED USING BINARY REPRESENTATION.

$\hat{x}_1$ - 00
$\hat{x}_2$ - 01
$\hat{x}_3$ - 10
$\hat{x}_4$ - 11 AS AN EXAMPLE

FOR $N = 2^R$ LEVELS, REQUIRE R BITS OR $R = \log_2 N$ BITS