Berkeley
UNIVERSITY OF CALIFORNIA

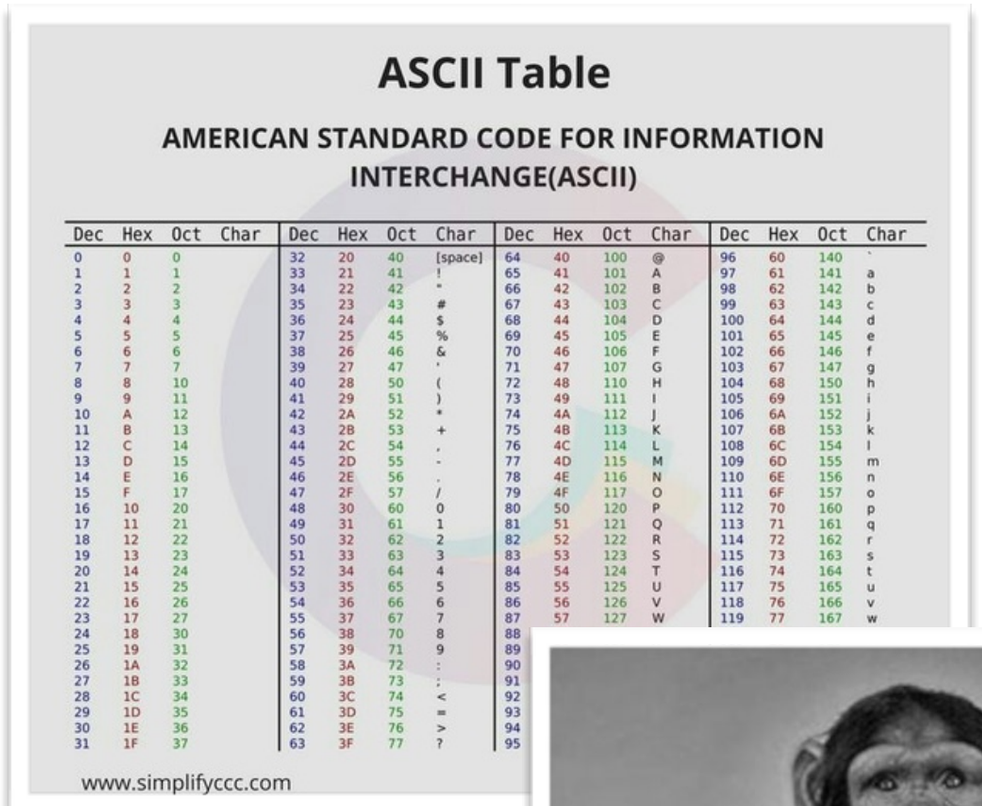
CNMAT

CARMINE-EMANUELE CELLA

INTRODUCTION TO DIGITAL SIGNALS

MUSIC 159

COMPUTERS ONLY GET NUMBERS...

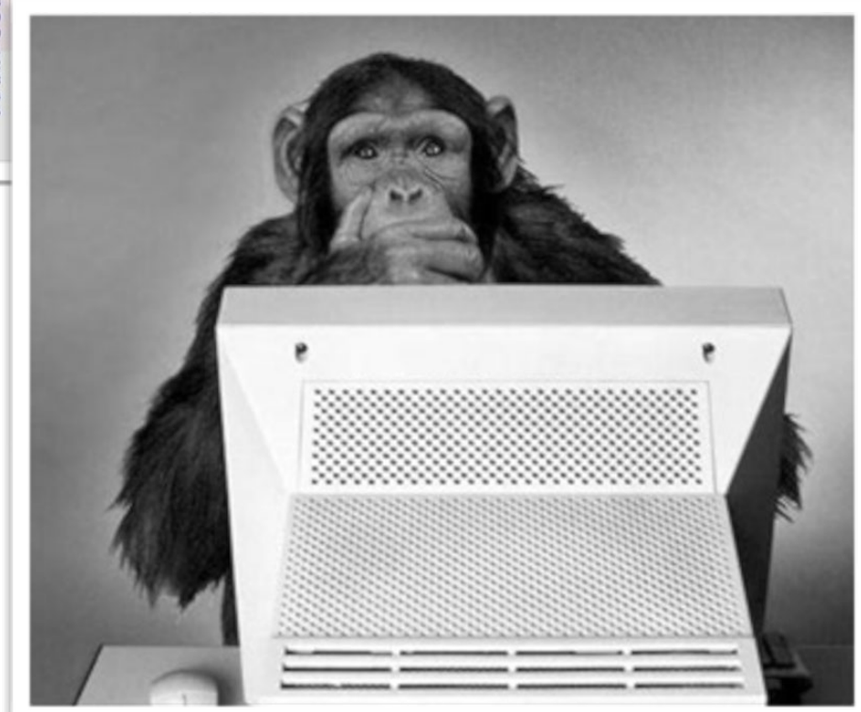


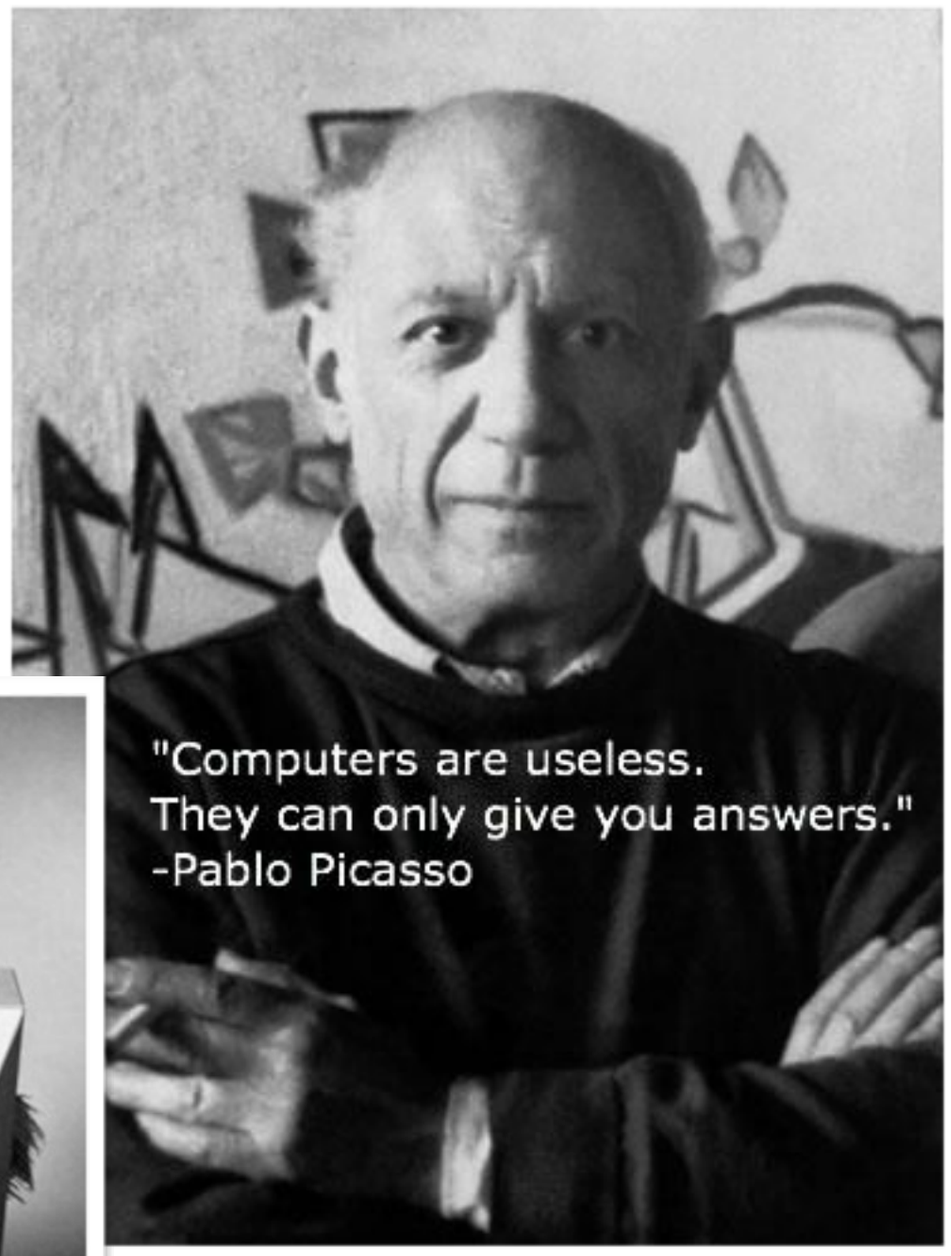
ASCII Table

AMERICAN STANDARD CODE FOR INFORMATION INTERCHANGE(ASCII)

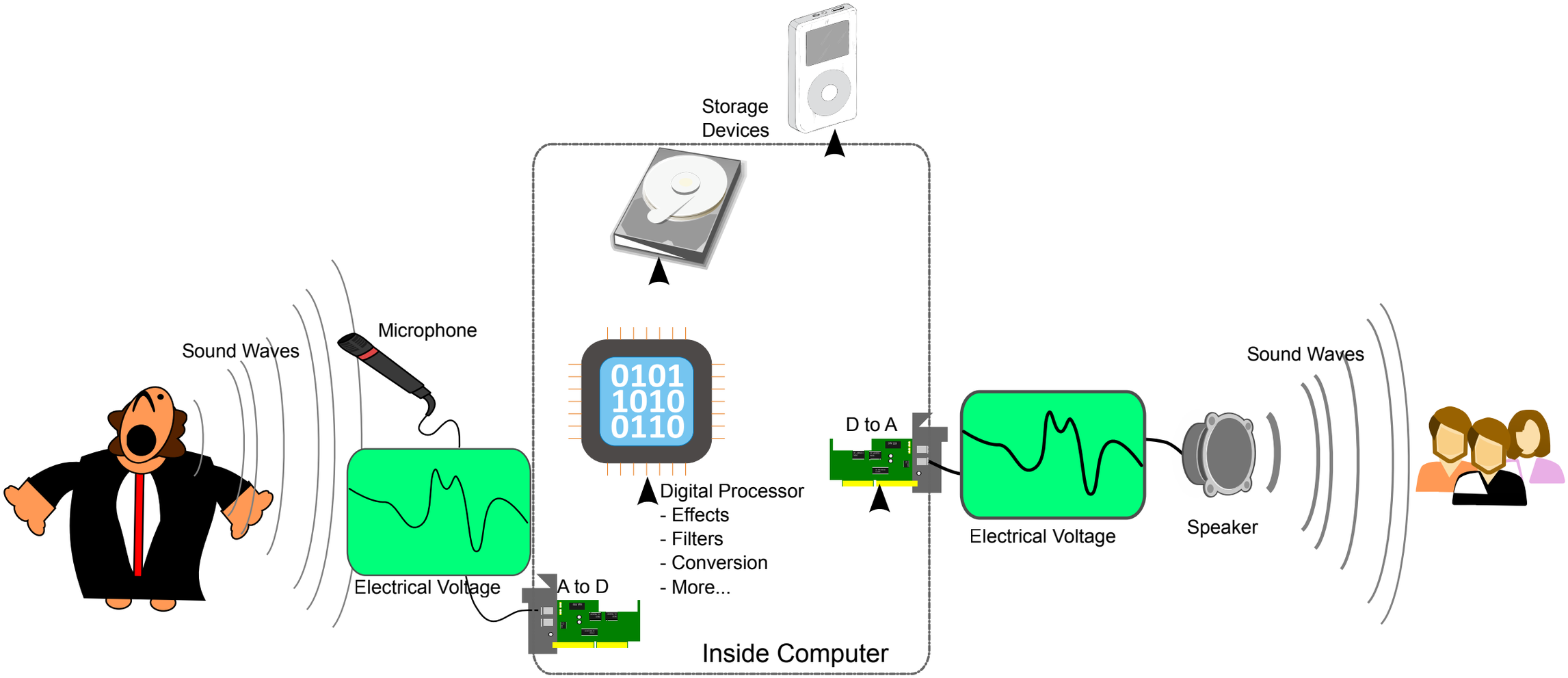
Dec	Hex	Oct	Char	Dec	Hex	Oct	Char	Dec	Hex	Oct	Char	Dec	Hex	Oct	Char
0	0	0		32	20	40	[space]	64	40	100	@	96	60	140	`
1	1	1		33	21	41	!	65	41	101	A	97	61	141	a
2	2	2		34	22	42	"	66	42	102	B	98	62	142	b
3	3	3		35	23	43	#	67	43	103	C	99	63	143	c
4	4	4		36	24	44	$	68	44	104	D	100	64	144	d
5	5	5		37	25	45	%	69	45	105	E	101	65	145	e
6	6	6		38	26	46	&	70	46	106	F	102	66	146	f
7	7	7		39	27	47	'	71	47	107	G	103	67	147	g
8	8	10		40	28	50	(	72	48	110	H	104	68	150	h
9	9	11		41	29	51	)	73	49	111	I	105	69	151	i
10	A	12		42	2A	52	*	74	4A	112	J	106	6A	152	j
11	B	13		43	2B	53	+	75	4B	113	K	107	6B	153	k
12	C	14		44	2C	54	,	76	4C	114	L	108	6C	154	l
13	D	15		45	2D	55	-	77	4D	115	M	109	6D	155	m
14	E	16		46	2E	56	.	78	4E	116	N	110	6E	156	n
15	F	17		47	2F	57	/	79	4F	117	O	111	6F	157	o
16	10	20		48	30	60	0	80	50	120	P	112	70	160	p
17	11	21		49	31	61	1	81	51	121	Q	113	71	161	q
18	12	22		50	32	62	2	82	52	122	R	114	72	162	r
19	13	23		51	33	63	3	83	53	123	S	115	73	163	s
20	14	24		52	34	64	4	84	54	124	T	116	74	164	t
21	15	25		53	35	65	5	85	55	125	U	117	75	165	u
22	16	26		54	36	66	6	86	56	126	V	118	76	166	v
23	17	27		55	37	67	7	87	57	127	W	119	77	167	w
24	18	30		56	38	70	8	88							
25	19	31		57	39	71	9	89							
26	1A	32		58	3A	72	:	90							
27	1B	33		59	3B	73	;	91							
28	1C	34		60	3C	74	<	92							
29	1D	35		61	3D	75	=	93							
30	1E	36		62	3E	76	>	94							
31	1F	37		63	3F	77	?	95							

www.simplifyccc.com





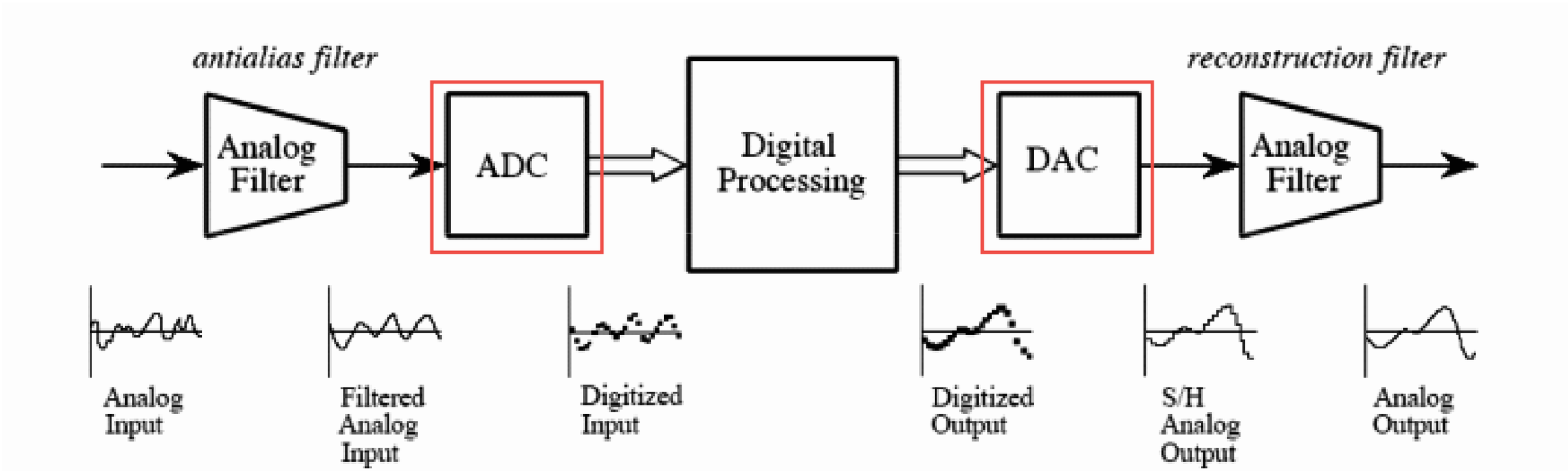
ANALOG/DIGITAL CONVERSION



in ⟶ out

ANALOG/DIGITAL CONVERSION (AGAIN)

The analog/digital conversion is done by the ADC and the DAC

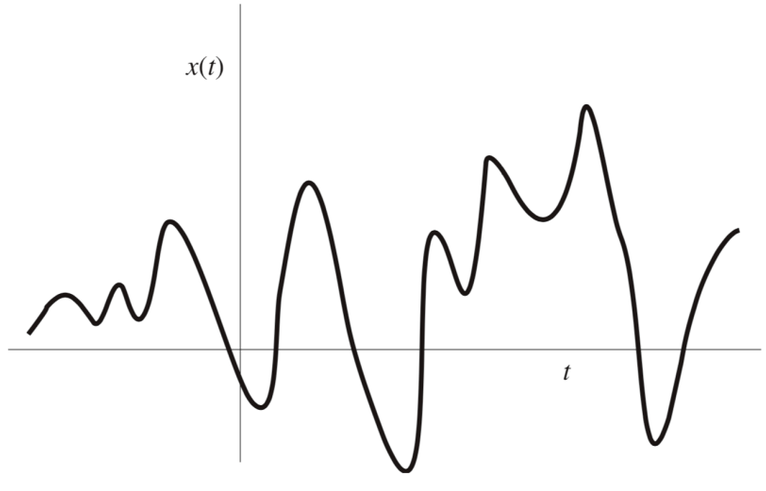


Mmm ..that's boring :(

CONTINUOUS TIME SIGNALS

- A *continuous-time signal* is a signal that is defined at each and every instant of time.
- Typical examples are:
  - An electromagnetic wave originating from a distant galaxy
  - The sound wave produced by a dolphin
  - The ambient temperature
  - The light intensity along the x and y axes in a photograph
- A continuous-time signal can be represented by a function

$$x(t) \quad \text{where} \quad -\infty < t < \infty$$



https://www.ece.uvic.ca/~dsp/support-materials.html

DISCRETE TIME SIGNALS (1)

- A *discrete-time signal* is a signal that is defined at discrete instants of time.
- Typical examples are:
  - The closing price of a particular commodity on the stock exchange
  - The daily precipitation
  - The daily temperature of a patient as recorded by a nurse

https://www.ece.uvic.ca/~dsp/support-materials.html

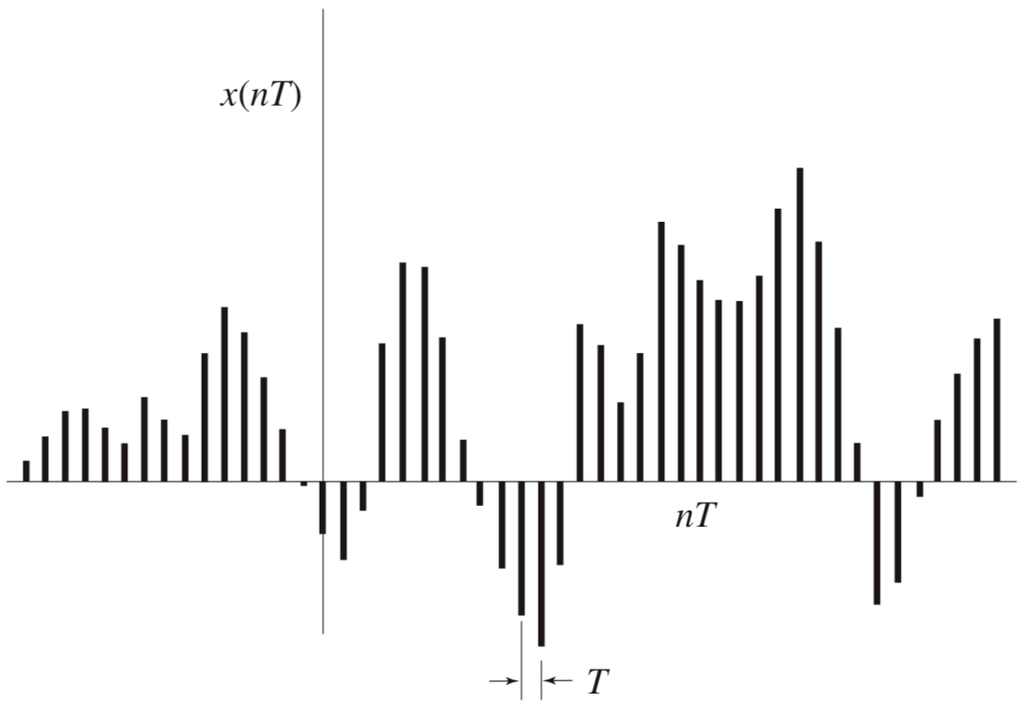
DISCRETE TIME SIGNALS (2)

- A discrete-time signal can be represented as a function

$$x(nT) \quad \text{where} \quad -\infty < n < \infty$$

  and T is a constant.

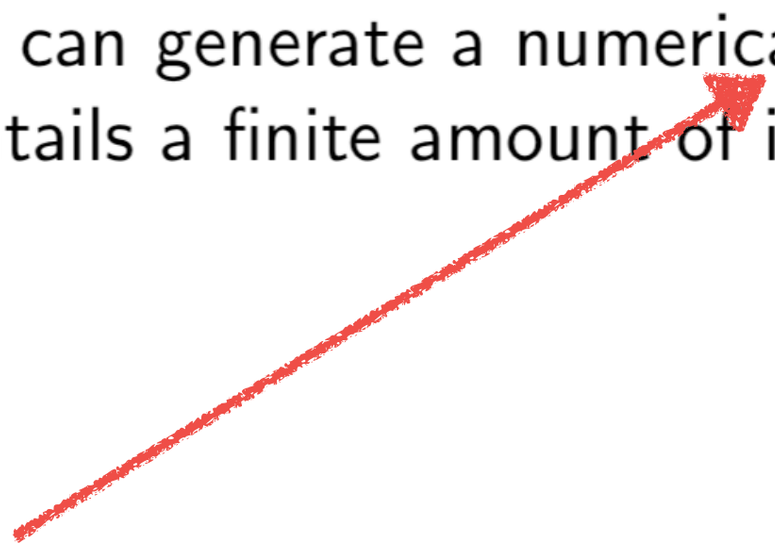
- The quantity $x(nT)$ can represent a voltage or current level or any other quantity.



https://www.ece.uvic.ca/~dsp/support-materials.html

SAMPLING

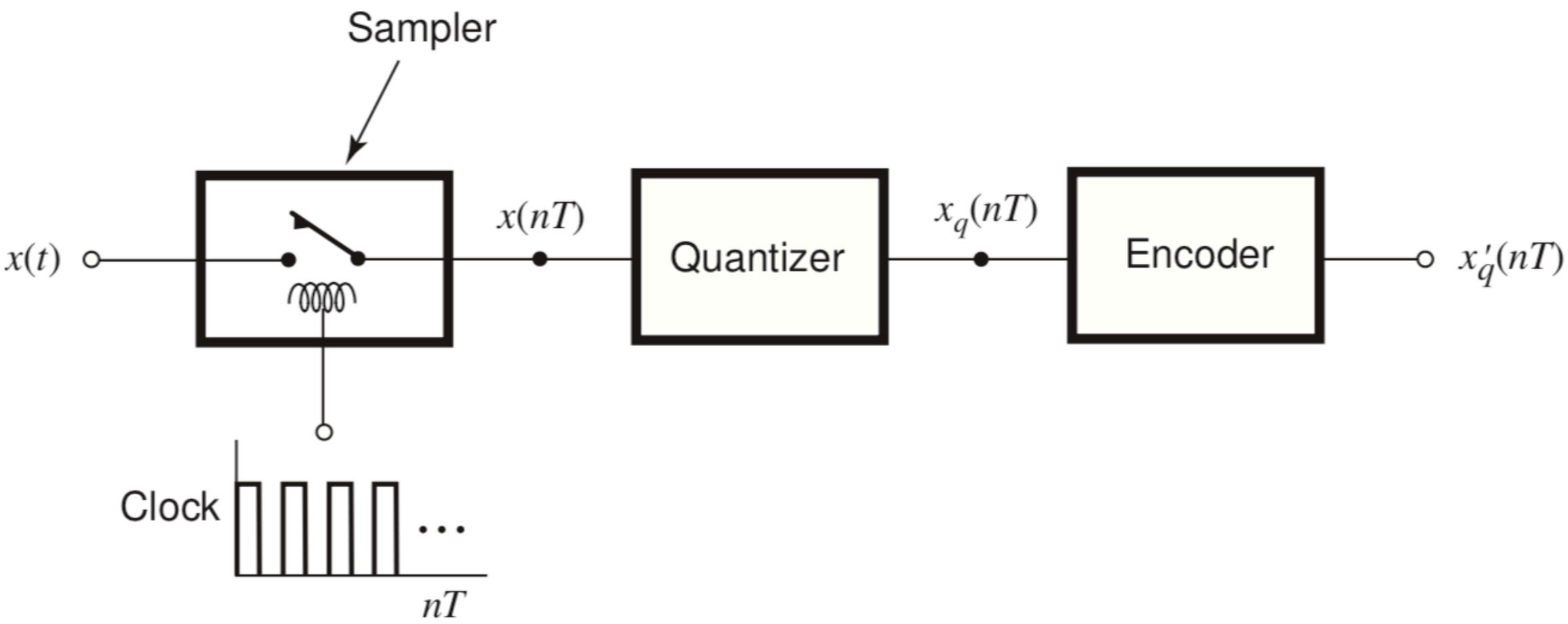
- To be able to process a nonquantized continuous-time signal by a digital system, we must first sample it to generate a discrete-time signal.
- We must then quantize it to get a quantized discrete-time signal.
- That way, we can generate a numerical representation of the signal that entails a finite amount of information.



The output numbers are called "samples"

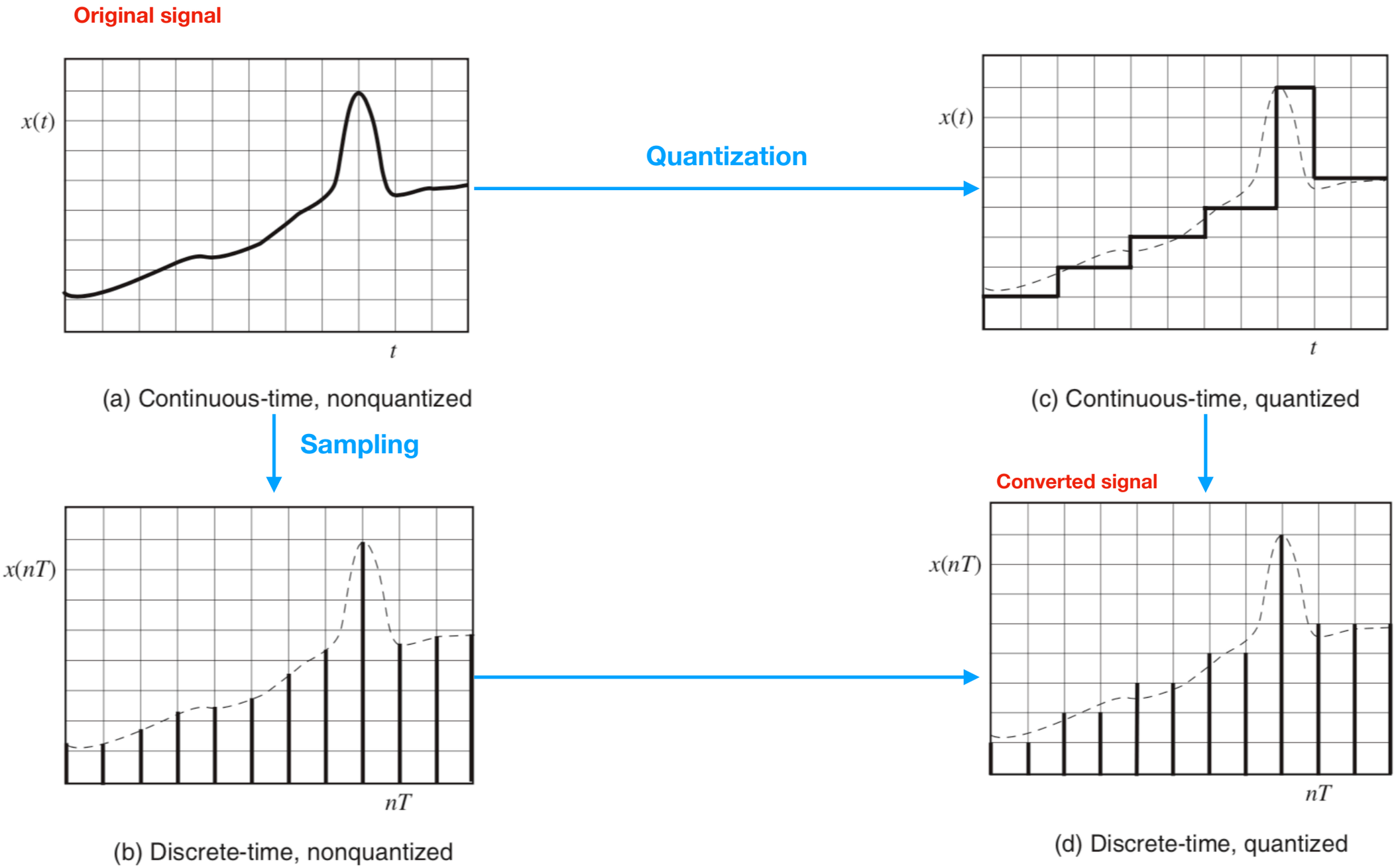
https://www.ece.uvic.ca/~dsp/support-materials.html

ADC (ANALOG-TO-DIGITAL CONVERTER)



https://www.ece.uvic.ca/~dsp/support-materials.html

JOINT ACTION OF SAMPLING AND QUANTIZATION



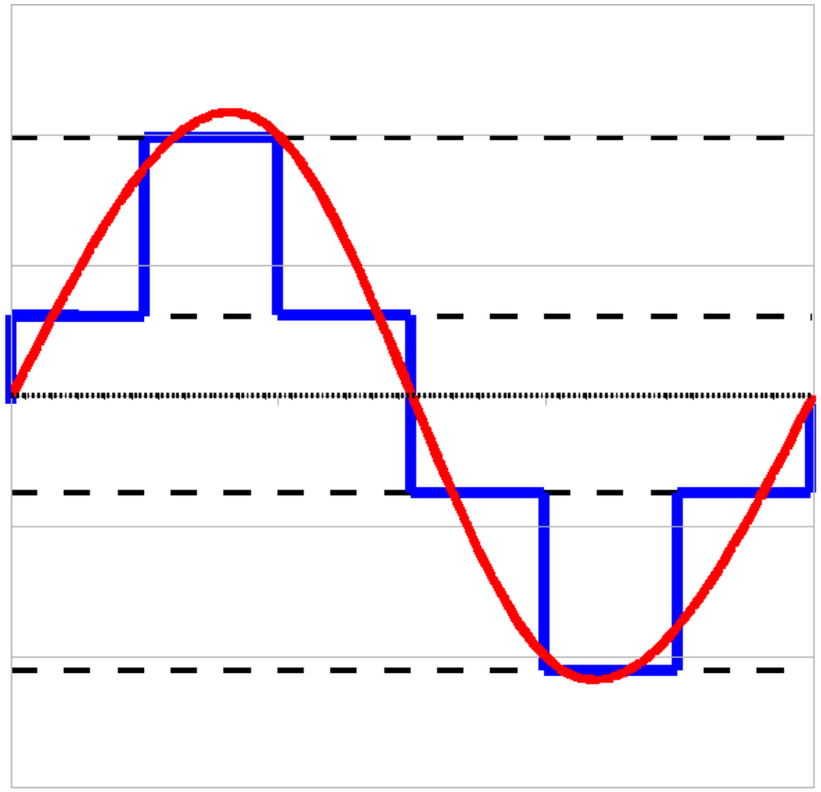
https://www.ece.uvic.ca/~dsp/support-materials.html

SAMPLING RATE (SR)

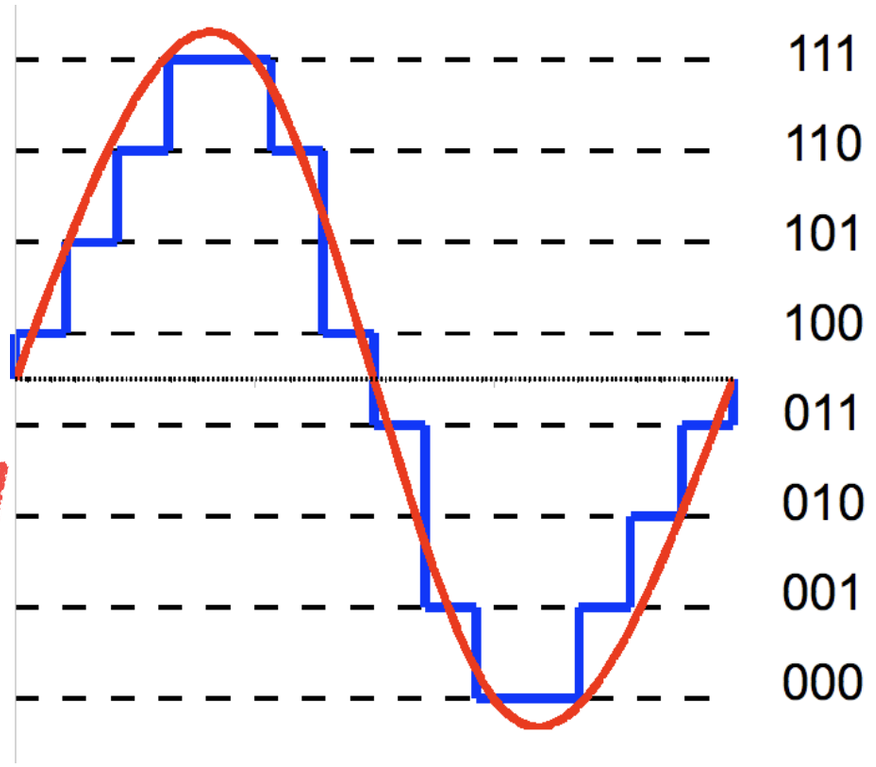
- A quantized discrete-time signal produced by an A/D converter is, of course, an approximation of the original nonquantized continuous-time signal.
- The accuracy of the representation can be improved by increasing
  - the sampling rate, and/or
  - the number of allowable quantization levels in the quantizer
- The sampling rate is simply $1/T = f_s$ in Hz or $2\pi/T = \omega_s$ in radians per second (rad/s).

The sampling rate (SR) is measured in Hz, while the quantization levels are measured in bits; can you tell why?

QUANTIZATION LEVELS



11
10
01
00

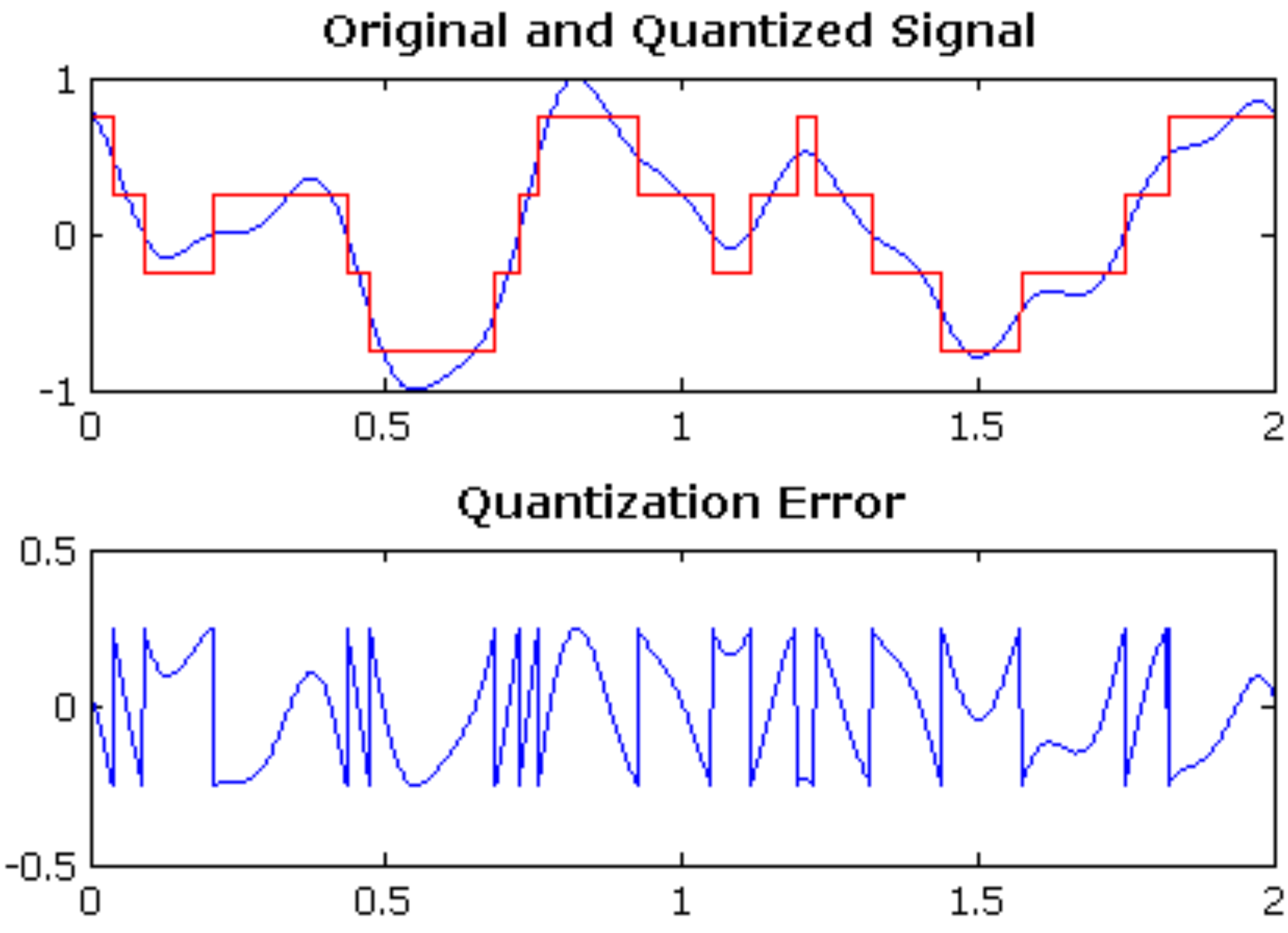


2-bit quantization

3-bit quantization

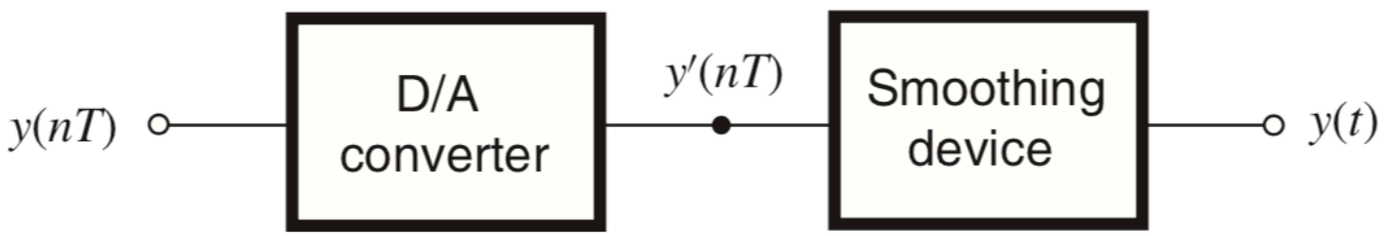
Quantization levels represent the dynamic range of a signal

QUANTIZATION ERROR



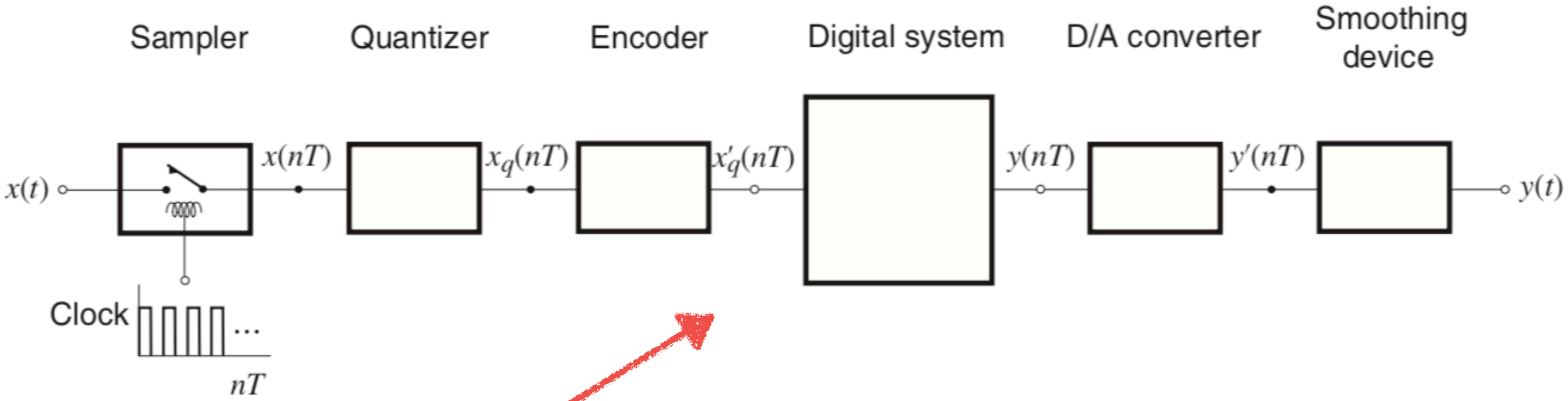
DAC (DIGITAL-TO-ANALOG CONVERTER)

- If the processed discrete-time signal is intended for a person, e.g., a music signal, then it must be converted back into a continuous-time signal.
- Just like the sampling process, the conversion from a discrete- to a continuous-signal requires a suitable *digital-to-analog interface*.
- Typically, the digital-to-analog interface requires a series of two cascaded modules, a *digital-to-analog (or D/A) converter* and a *smoothing device:*



https://www.ece.uvic.ca/~dsp/support-materials.html

ANALOG/DIGITAL CONVERSION (FINALLY)



Mmm ..that's REALLY boring :(

https://www.ece.uvic.ca/~dsp/support-materials.html

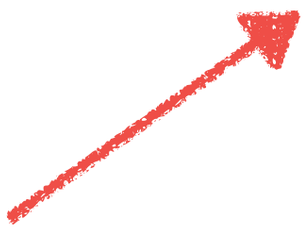
SAMPLING THEOREM (SHANNON-NYQUIST)

- **Discovered by Claude Shannon in 1949:**

A signal can be reconstructed from its samples without loss of information, if the original signal has no frequencies above 1/2 the sampling frequency

Fs >= 2 * f

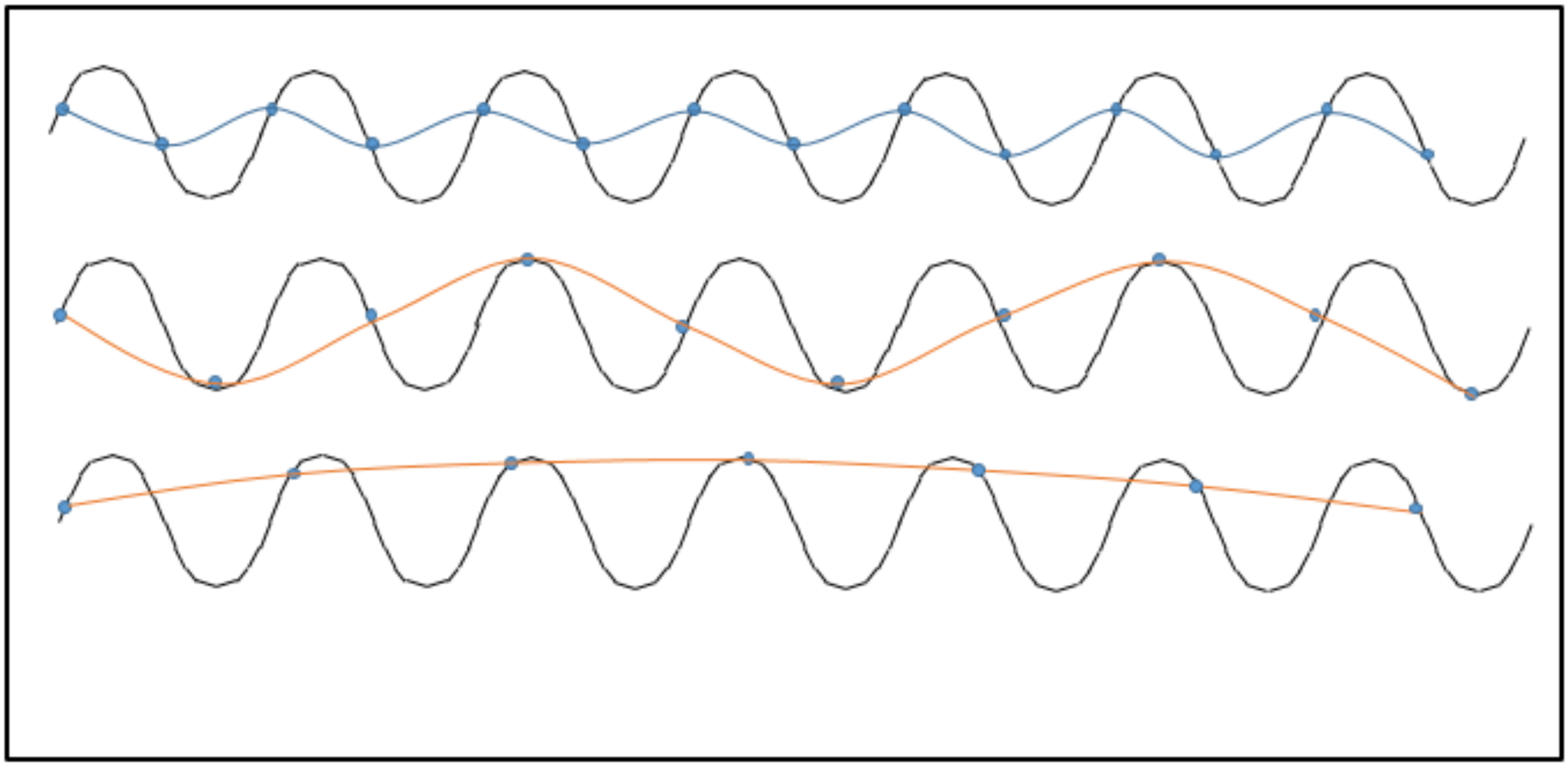
When the signal is sampled with a sampling rate that is lower than double of its highest frequency, aliasing occurs



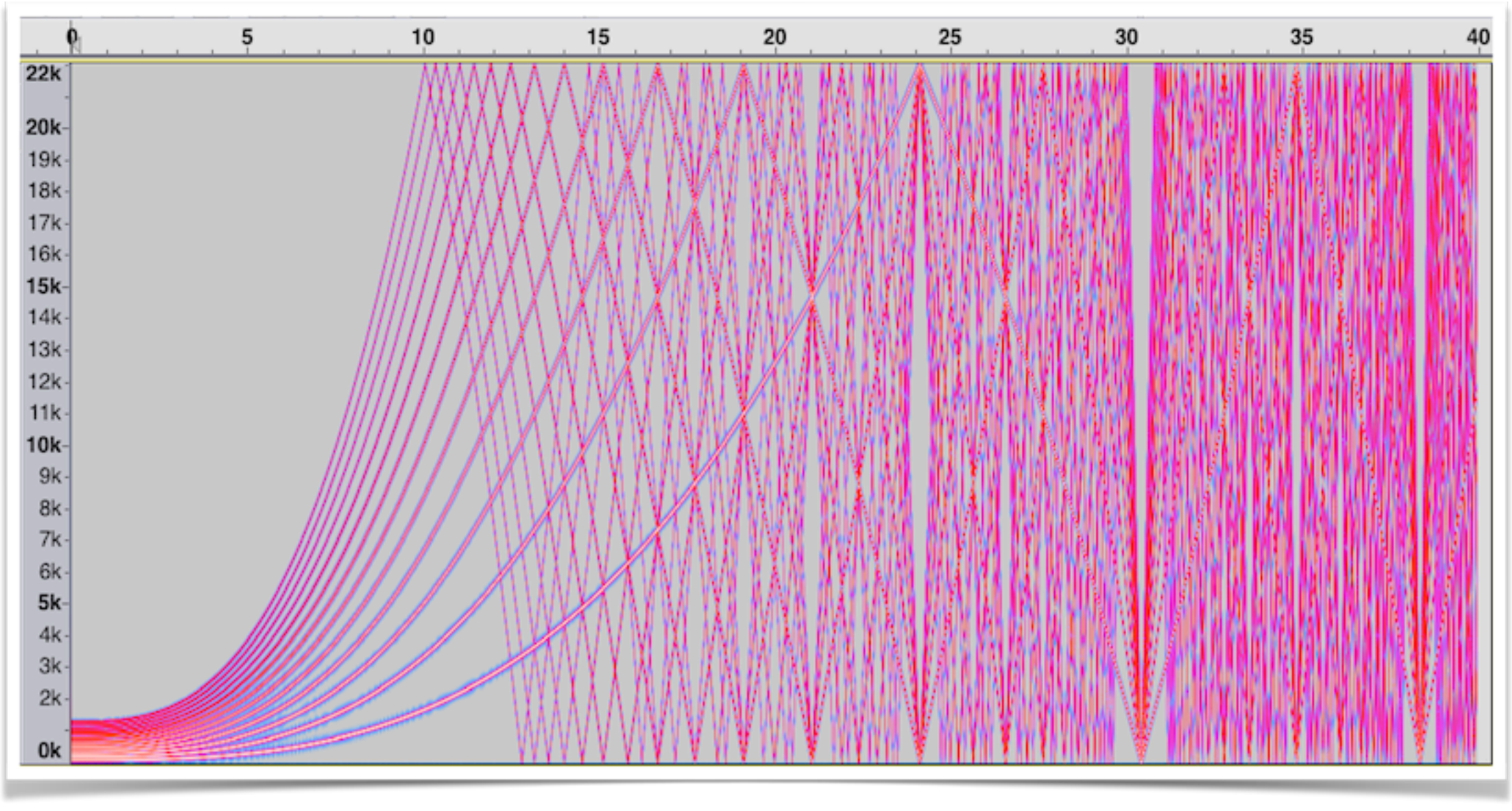
The theorem states that you need at least two samples per period

ALIASING (WRONG SAMPLING)

In aliasing, the frequencies *F* in the signal that are higher than 1/2 the sampling rate *SR*, are transformed to low frequencies *LS = F - 1/2 SR*



ALIASING (EXAMPLE USING SONOGRAM)

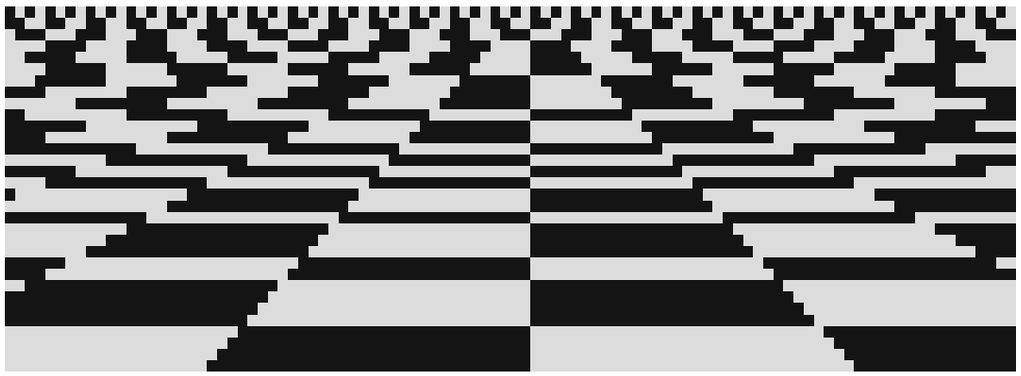


https://sites.math.washington.edu/~conroy/sequenceNoise/aliasingExamples.htm

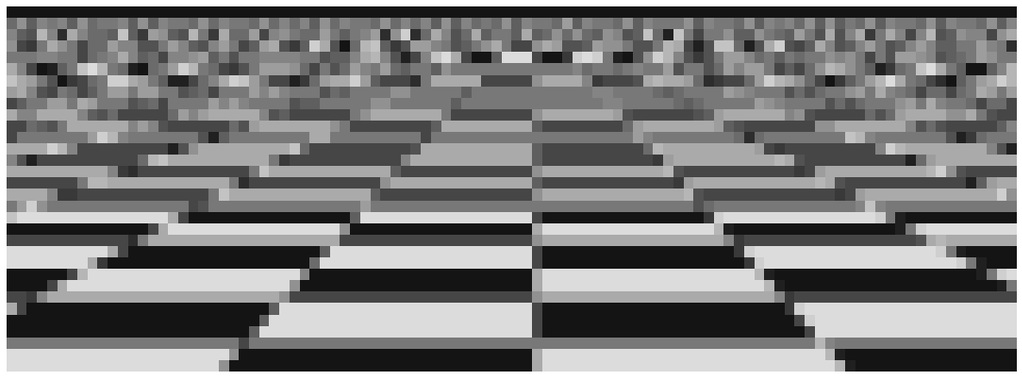
SPATIAL ALIASING (1)

https://en.wikipedia.org/wiki/Aliasing

SPATIAL ALIASING (2)



Point sampling



4x4 Supersampling

AVOIDING ALIASING

1. **Increase sampling rate**
   - ➢ **Not a general-purpose solution**
     - **White noise is not band-limited**
     - **Faster sampling requires:**
       - – **Faster ADC**
       - – **Faster CPU**
       - – **More power**
       - – **More RAM for buffering**
2. **Filter out undesirable frequencies before sampling using analog filter(s)**
   - ➢ **This is what is done in practice**
   - ➢ **Analog filters are imperfect and require tradeoffs**

DIGITAL CD STANDARD

- Sampling Rate: 44.1 kHz
- 16-bit samples
- 2-channel stereo
- Data transfer rate = 2×16×44,100 = **1.4 Mbits/s**
- 1 hour of music = 1.4×3,600 = **635 MB**

Is it better than a vinyl??

Question: which are the best sampling rate and quantization levels to record your guitar?

SUMMARY (SHORT VIDEO)

Nyquist Shannon Sampling Theorem

Group5: Cian and Carren

Questions:

Humans can hear up to about 20KHz; this means, by the Shannon-Nyquist theorem, that a sampling rate of 40 KHz would be enough.

Why the CD goes up to 44100 Hz?

Is there any need to record sound at 96 KHz?

SIGNAL PROCESSING

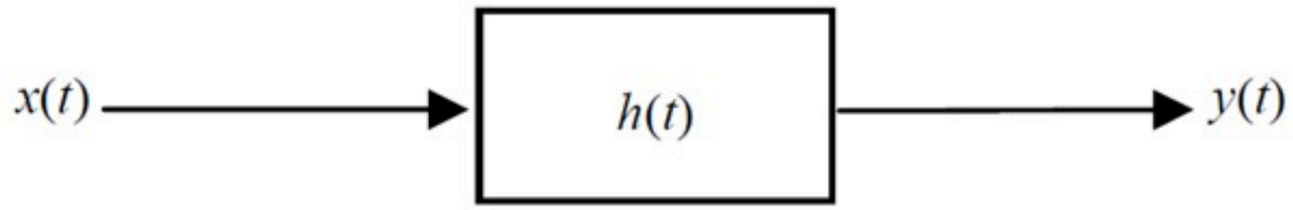
- Signal processing emerged soon after World War I in the form of electrical filtering.
- With the invention of the digital computer and the rapid advances in VLSI technology during the 1960s, a new way of processing signals emerged: *digital signal processing*.

(Ouch... I thought that the class was finished)

DIGITAL SIGNAL PROCESSING (DSP)

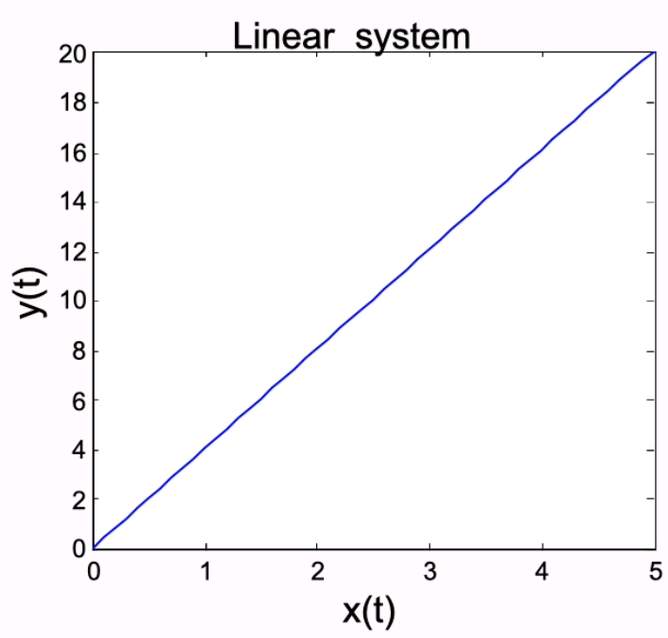
- Signal processing is the science of analyzing, synthesizing, sampling, encoding, transforming, decoding, enhancing, transporting, archiving, and generally manipulating signals in some way or another.
- These presentations are concerned primarily with the branch of signal processing that entails the manipulation of the *spectral characteristics of signals.*
- If the processing of a signal involves modifying, reshaping, or transforming the spectrum of the signal in some way, then the processing involved is usually referred to as *filtering*.
- If the filtering is carried out by digital means, then it is referred to as *digital filtering*.

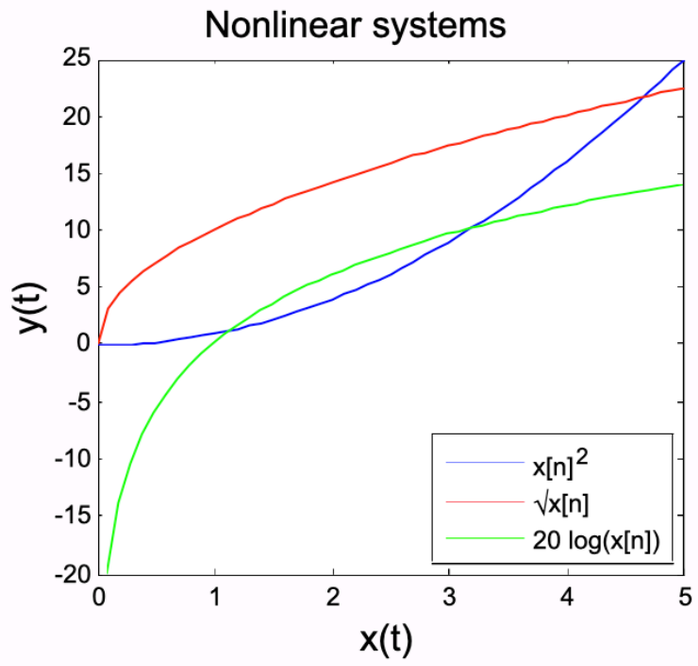
LTI: LINEAR-TIME INVARIANT SYSTEMS (1)



- What is a Linear system:
  - The system applies to superposition

$$T\{a\,x_1(t)+b\,x_2(t)\}=a\,T\{x_1(t)\}+b\,T\{x_2(t)\}$$





LTI: LINEAR-TIME INVARIANT SYSTEMS (2)

- Time invariant:
- A time invariant systems is independent on explicit time
  - (The coefficient are independent on time)
- That means

If: $y_2(t)=f[x_1(t)]$

Then: $y_2(t+t_0)=f[x_1(t+t_0)]$

The same to Day tomorrow and in 1000 years

EXAMPLES

- A linear system

  $y(t)=3\,x(t)$

- A nonlinear system

  $y(t)=3\,x(t)^2$

- A time invariant system

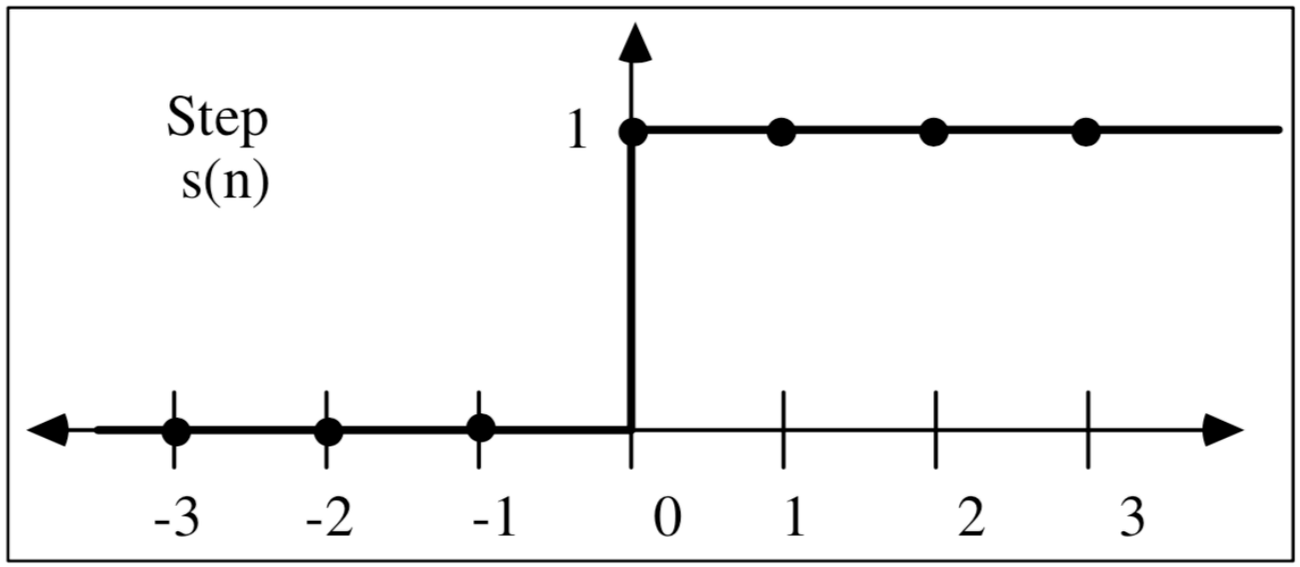
  $y(t)=3\,x(t)$

- A time variant system

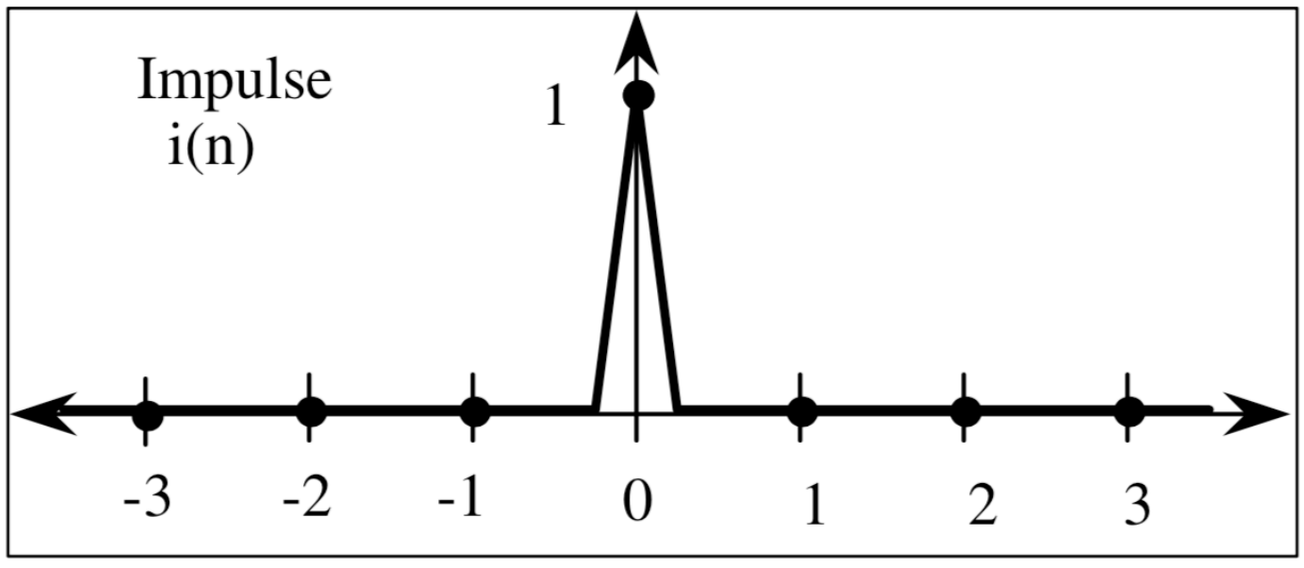
  $y(t)=3^t\,x(t)$

IMPORTANT SIGNALS IN LTI SYSTEMS

- **Step:**
  - **…, 0, 0, 0, 1, 1, 1, …**



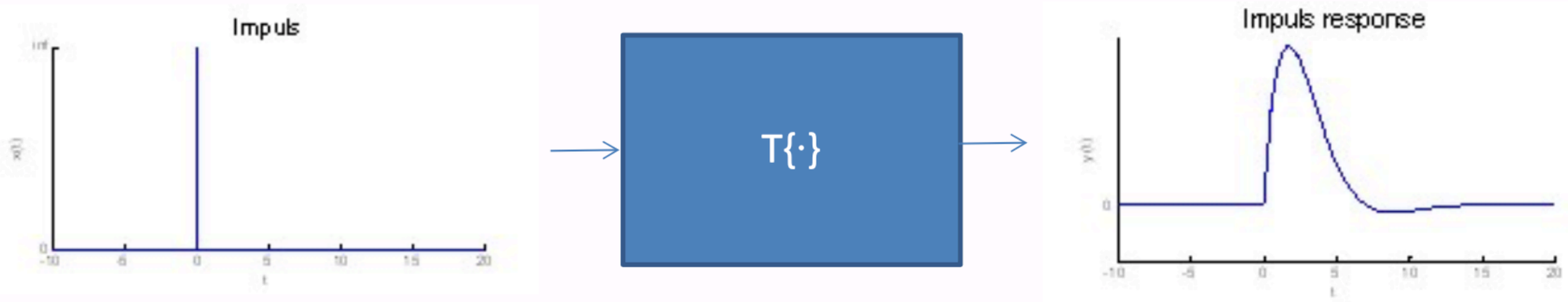
- **Impulse:**
  - **…, 0, 0, 0, 1, 0, 0, …**



IMPULSE RESPONSE

The output of a system if Dirac delta is input

$\delta(t)$ $\quad h[n] = f[\delta(t)] \quad$ $h(t)$



GEEK'S CORNER – 8 BIT MUSIC GENERATION IN C ONE-LINERS

```c
int main (t) {
    for (;;t++) {
        putchar((t*9&t>>4|t*5&t>>7|t*3&t/1024)-1);
    }
}
```

```c
int main(int t) {
    for(;;t++)putchar(t&t%255 - (t*3&t>>13&t>>6));
    return 0;
}
```

THANK YOU!

Suggested exercise:
experiment changing sampling rate and quantization in a DAW!