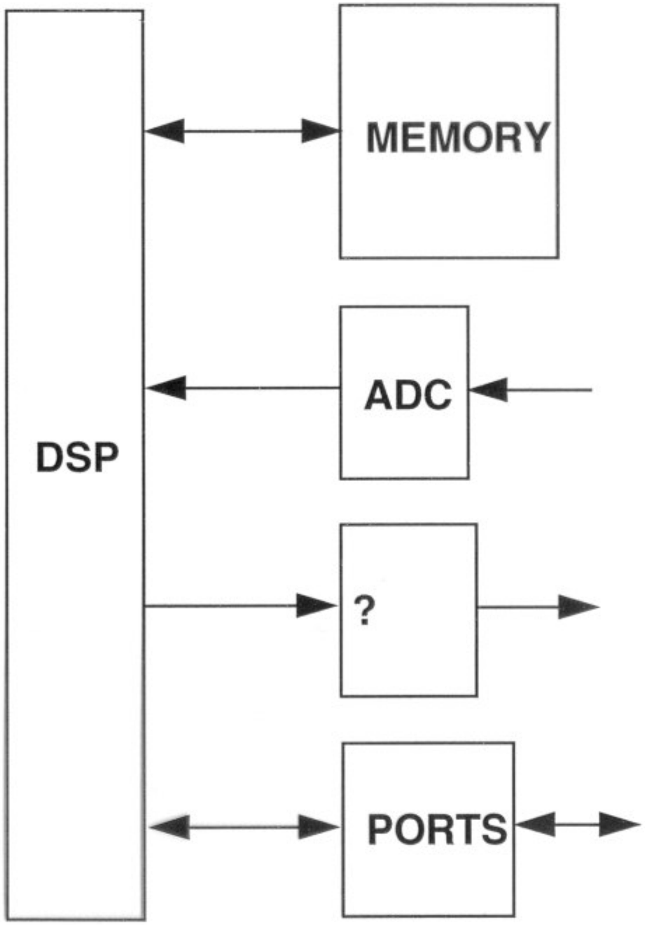
A TYPICAL DSP SYSTEM

Typické usporiadanie systému „Digital Signal Processor“



- **DSP CHIP** Číslicový signálový procesor
- **MEMORY**
- **CONVERTERS (OPTIONAL)** Prevodníky (voliteľná súčasť)
  - ANALOGUE TO DIGITAL
  - DIGITAL TO ANALOGUE
- **COMMUNICATION PORTS**
  - SERIAL
  - PARALLEL

MULTIPLY AND ADD

Operácie násobenia (Multiply) a sčítania (Add)

ADD

1+2=3

```
  0001
+ 0010
------
  0011
```

MULTIPLY

2*4=8

0010

SHIFT LEFT TWICE

posuň dvakrát doľava

?

Najviac používaná operácia v DSP

MOST COMMON OPERATION IN DSP

$\boxed{A = B*C + D}$

E = F*G + A

⋮

MULTIPLY, ADD AND ACCUMULATE

MAC INSTRUCTION

násob a výsledok pričítaj k predošlému výsledku (operand A)

MULTIPLY OPERATION

TYPICALLY 70 CLOCK CYCLES WITH ORDINARY PROCESSORS

v bežnom mikroradiči trvá výpočet násobenia až 70 strojových cyklov

WE NEED TO MAC IN ONE CYCLE

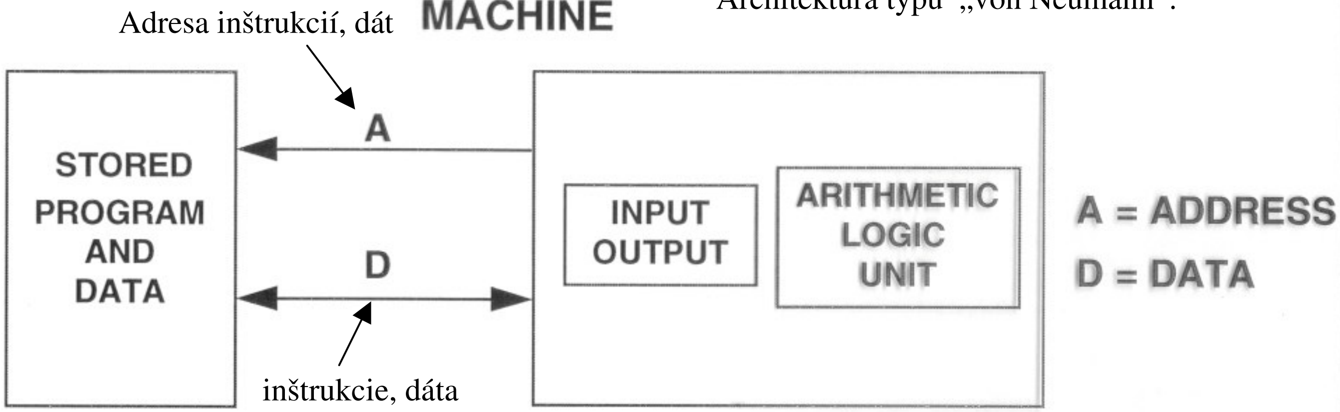
My ale potrebujeme násobiť a pripočítavať v JEDNOM strojovom cykle

DIGITAL COMPUTERS

Procesory - dve architektúry:

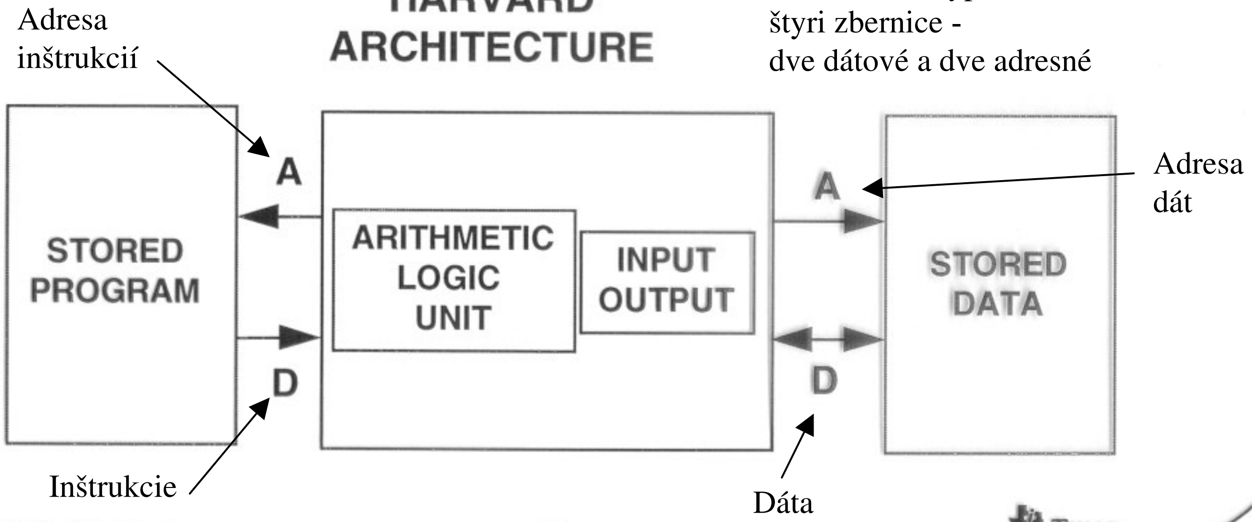
VON NEUMAN MACHINE

Architektúra typu „von Neumann“:



HARVARD ARCHITECTURE

Architektúra typu „Harvard“:
štyri zbernice -
dve dátové a dve adresné



WHY DIGITAL?

Prečo číslicové spracovanie?

- IS IT WORTH IT?

Stojí to zato? Zaslúži si to?

- IS DIGITAL PROCESSING BETTER?

Je číslicové spracovanie výhodnejšie?

  - APPLICATION DEPENDENT
  - PROGRAMMABILITY
  - STABILITY AND REPEATABILITY
  - SPECIAL APPLICATIONS

Niekedy áno niekedy nie, závisí na aplikácii

Programovateľnosť, zmena programu - zmena funkcie

Lepšia časová a teplotná stabilita
dva rovnaké DSP systémy vykazujú viac rovnaké parametre

niektoré aplikácie je možné realizovať len digitálne

PROGRAMMABILITY

- ONE HARDWARE = MANY TASKS

Jedna schéma zapojenia - rôzne funkcie, závislé od programu

SOFTWARE 1 → SAME HARDWARE → LOW PASS FILTER
SOFTWARE 2 → SAME HARDWARE → MUSIC SYNTHESIZER
:
SOFTWARE N → SAME HARDWARE → MOTOR CONTROL

- UPGRADABILITY AND FLEXIBILITY

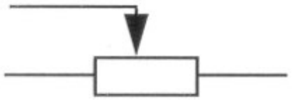
Pridanie funkcií a prispôsobivosť znamená:

  - DEVELOP NEW CODE → UPGRADE
  - ANALOGUE → SOLDER NEW COMPONENT

Číslicové sprac. - modifikovať program (do určitého stupňa nie je potrebné modifikovať zapojenie)
Analóg ové sprac. - modifikovať zapojenie

STABILITY AND REPEATABILITY

- ANALOGUE CIRCUITS ARE AFFECTED BY
  - TEMPERATURE
  - AGING



Na vlastnosti analógových obvodov vplýva teplota a starnutie súčiastok

- TOLERANCE OF COMPONENTS
  - TWO ANALOGUE SYSTEMS
    - USING SAME DESIGN
    - USING SAME COMPONENTS
    - MAY DIFFER IN PERFORMANCE

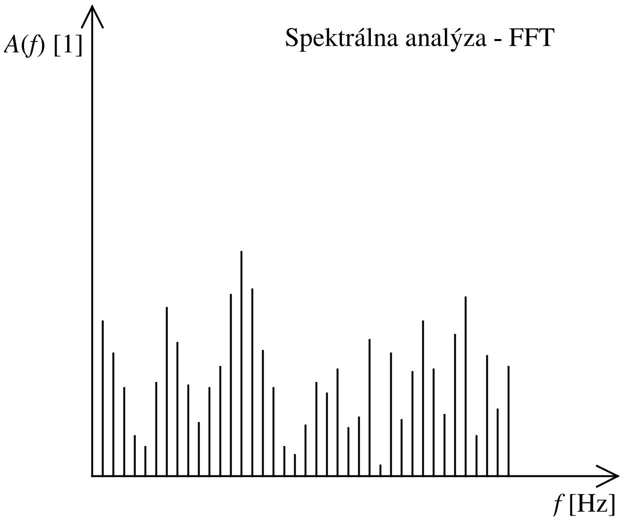
Rozptyl hodnôt súčiastok môže spôsobiť, že dve rovnaké zapojenia s rovnakými súčiastkami nemusia vykazovať rovnaké vlastnosti

PERFORMANCE

- SOME SPECIAL FUNCTIONS CAN ONLY BE IMPLEMENTED DIGITALLY Niektoré druhy spracovania signálov je možné realizovať iba číslicovo

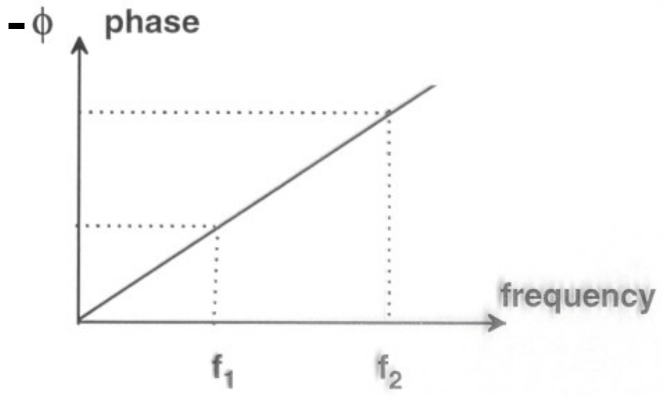
  - LOSSLESS COMPRESSION

Bezstratová a stratová kompresia zvukového a obrazového signálu - MPEG, ATRAC...



  - LINEAR PHASE FILTERS

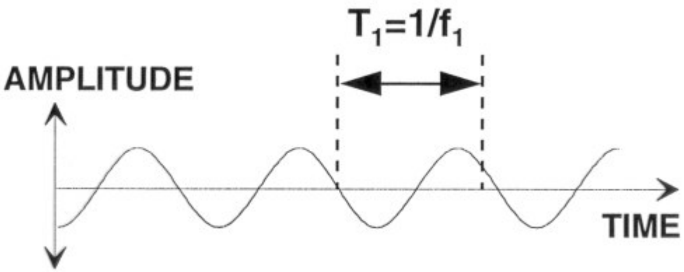
Filtre s lineárnou fázovo - frekvenčnou charakteristikou

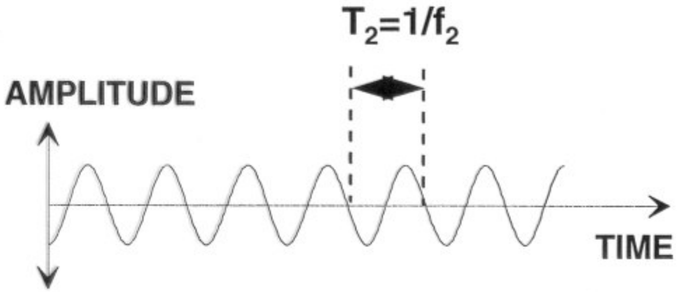


SIGNALS IN TIME AND FREQUENCY DOMAINS

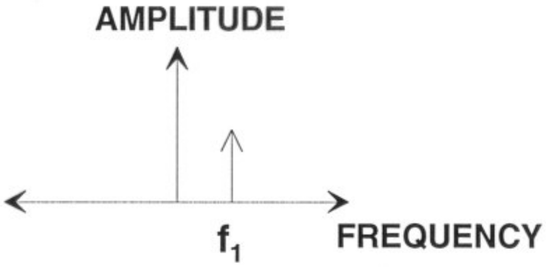
Signály v časovej a frekvenčnej oblasti

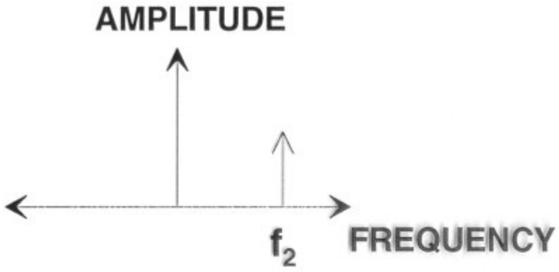
TIME DOMAIN





FREQUENCY DOMAIN

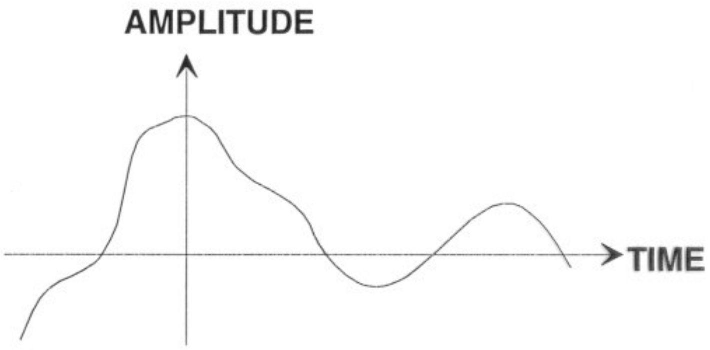




T = period   f = frequency

REAL SIGNALS

Reálne, všeobecné signály



Reálne, všeobecné signály sú zložené z mnohých harmonických signálov s rôznymi frekvenciami - frekvenčné zložky signálu

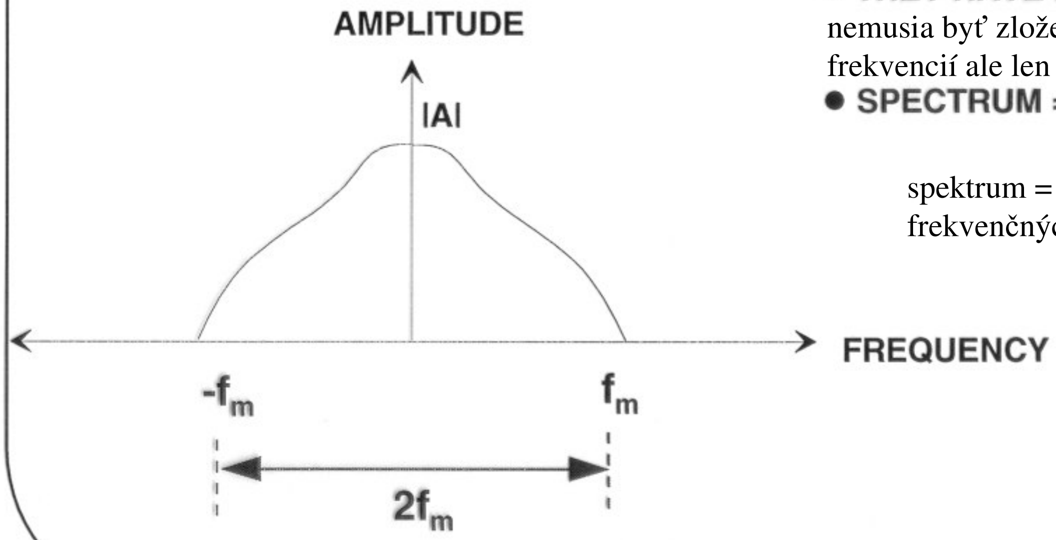
- REAL LIFE SIGNALS ARE A COMBINATION OF MANY FREQUENCIES

- THEY HAVE BANDWIDTH $2f_m$

nemusia byť zložené z harm. signálov všetkých frekvencií ale len z frekvencií z intervalu <0;*f*m>

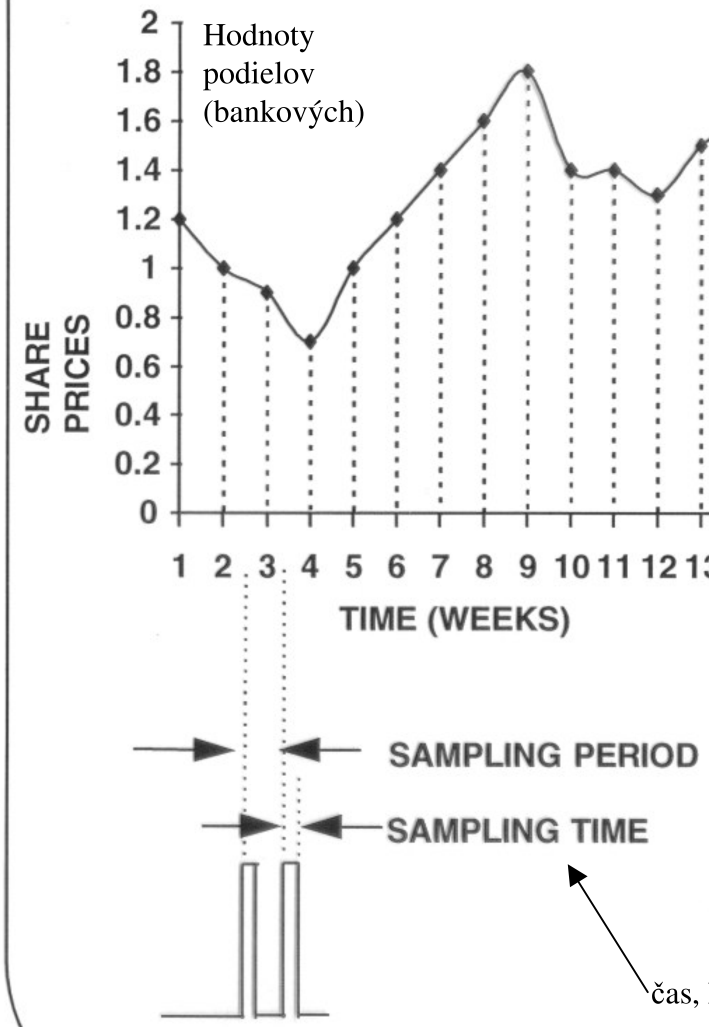
- SPECTRUM = FREQUENCY CONTENT

spektrum = zastúpenie jednotlivých frekvenčných zložiek



SAMPLING

vzorkovanie



- TAKE SNAPSHOTS OF CONTINUOUSLY CHANGING DATA

  urob fotky spojito sa meniacich hodnôt

- ‘SAMPLING PERIOD’ IS FIXED

  čas medzi dvoma „foteniami“ sa nazýva vzorkovacia perióda, má konštantnú hodnotu

- THIS MAKES INFORMATION UNDERSTANDABLE

  u týmto sa informácia stáva zmysluplnou (?)

- MY SHARE PRICE HIT ITS LOWEST IN WEEK 4

  hodnota mojich podielov dosiahla minimum vo štvrtom týždni

- MY SHARE PRICE REACHED ITS PEAK IN WEEK 9

  hodnota mojich podielov dosiahla maximum v deviatom týždni

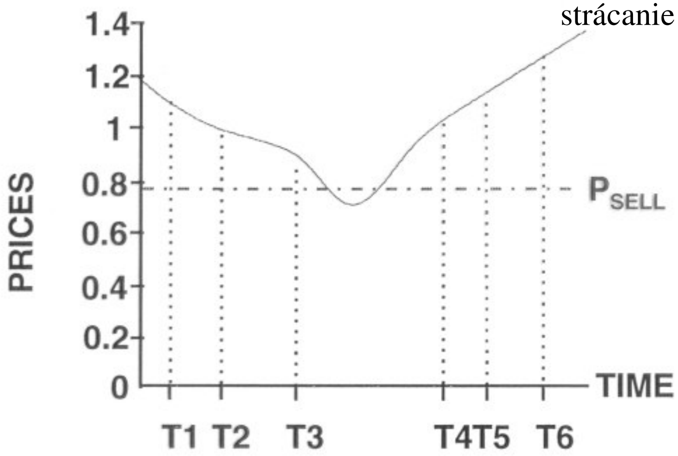
- SAMPLING PERIOD IS THE TIME BETWEEN SAMPLES

- SAMPLING TIME IS THE TIME TAKEN TO TAKE A SAMPLE, ‘A SNAPSHOT’

čas, ktorý je potrebný na „odfotenie“ hodnoty

MISSING INFORMATION

strácanie informácie



- **NON-PERIODIC SNAPSHOTS**

  nepravidelné „fotenie“

  - MAY MISS INFORMATION

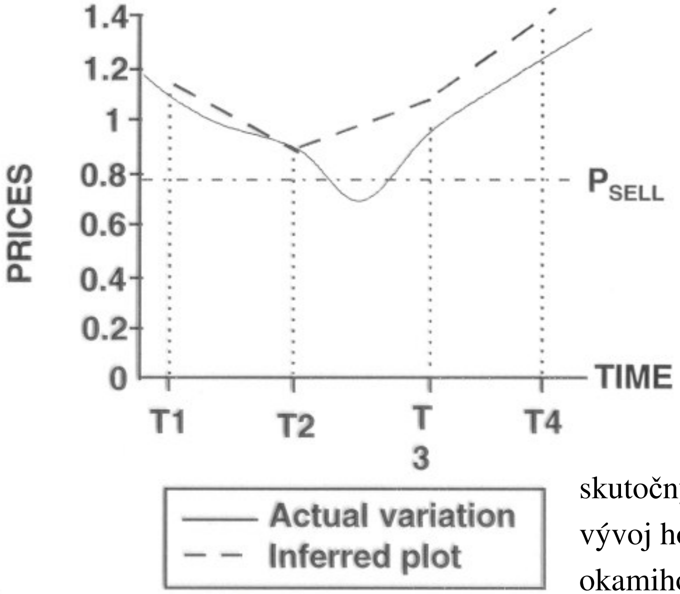
    môže dôjsť ku strate hodnoty

    THE DIP IN PRICES BETWEEN T3&T4 GOES UNNOTICED

    krátky pokles v hodnote medzi okamihmi T3 a T4 sa nezaznamenal

  - INFORMATION CANNOT BE INTERPRETED EASILY

    Informácia nemôže byť správne pochopená



- **PERIODIC SNAPSHOTS**

  pravidelné „fotenie“

  - EASIER TO INTERPRET

    Informácia môže byť lepšie pochopená

  - MAY STILL MISS INFORMATION

    stále ale môže dôjsť ku strate informácie

- THE KEY IS THE SAMPLING FREQUENCY

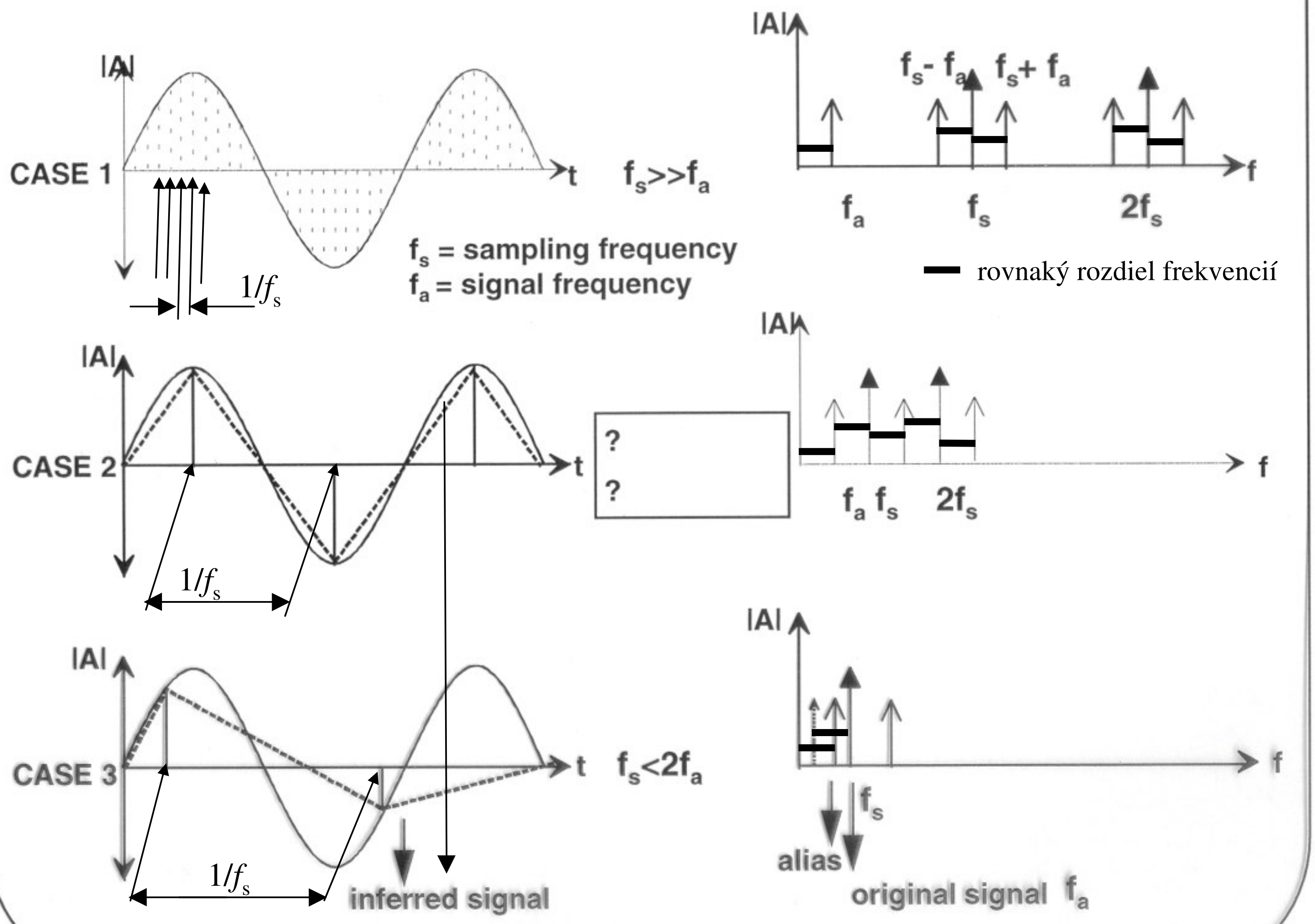
skutočný vývoj hodnoty

vývoj hodnoty signálu, jeho rekonštrukcia, ak poznáme jeho hodnoty iba v okamihoch T1 až T4

GETTING THE SAMPLING RIGHT

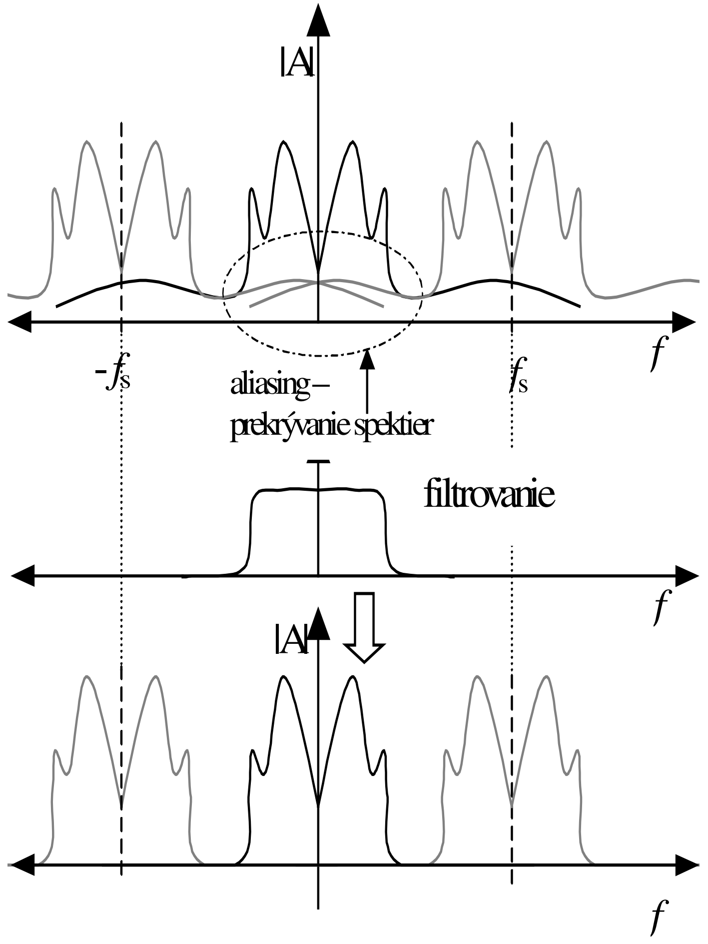
ako správne vzorkovať

TIME FREQUENCY



TEXAS INSTRUMENTS

LIMITING THE SPECTRUM



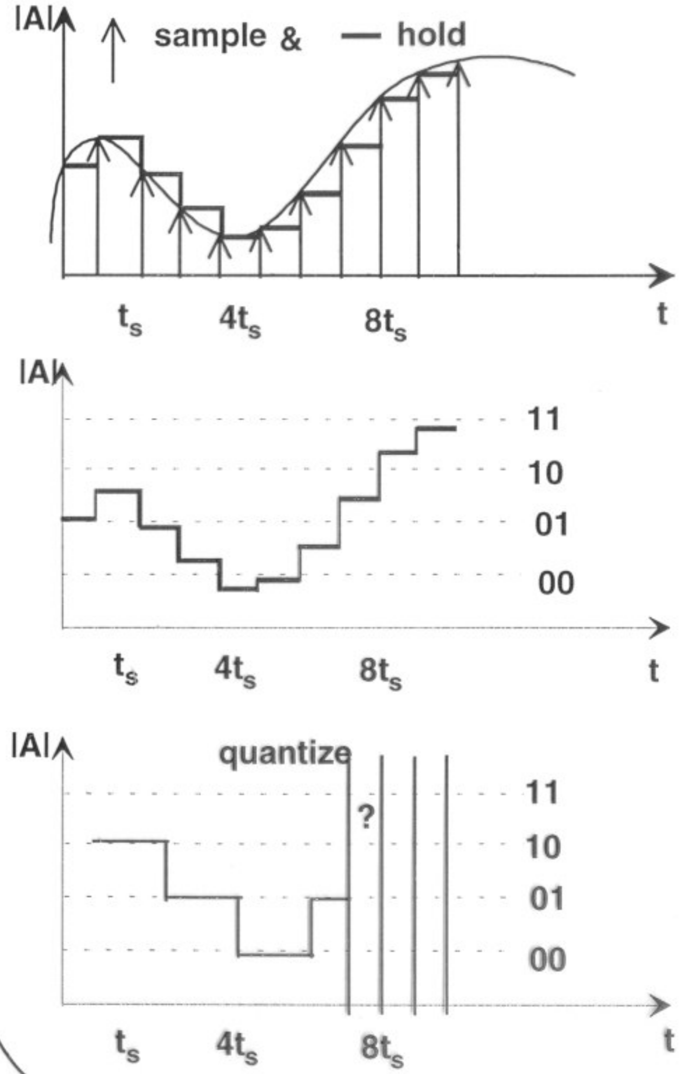
- SIGNALS IN THE REAL WORLD CONTAIN MANY FREQUENCIES
- FREQUENCY COMPONENTS GREATER THAN 1/2 f_s CAUSE ALIASING ($f > f_m$)

  frekvencie vyššie ako $f_s/2$ spôsobujú aliasing
- GET RID OF (=FILTER OUT) FREQUENCIES ABOVE f_m

  ako sa ich zbaviť - odfiltrovať ich
- THEN ENSURE SAMPLING RATE IS GREATER THAN $2f_m$

  a uistiť sa, že f_s je minimálne dvojnásobná ako je najvyššia frekvenčná zložka signálu

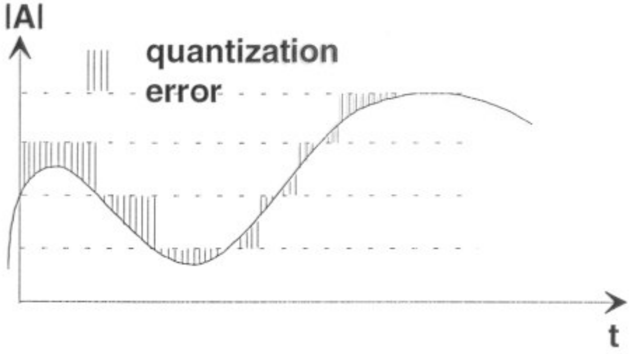
DIGITIZING THE SIGNAL



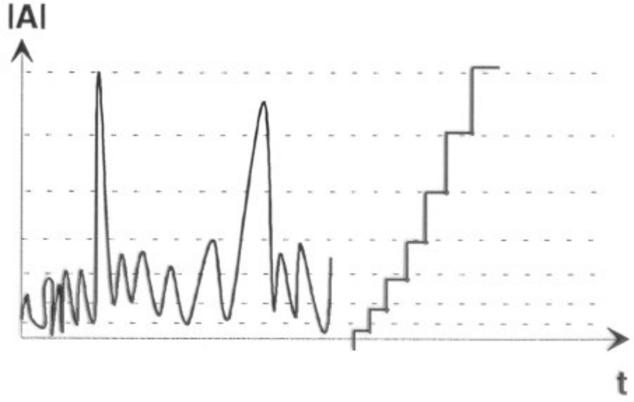
- AIM IS TO OBTAIN '1', '0' REPRESENTATION
- SAMPLE SIGNAL PERIODICALLY
- HOLD SAMPLED VALUE UNTIL NEXT SAMPLE
- CLASSIFY NEW SIGNAL INTO LEVELS = QUANTIZE
- MORE LEVELS, MORE ACCURATE

SIGNAL = 10 10 01 01 00 00 $\boxed{?}$

QUANTIZATION ERROR



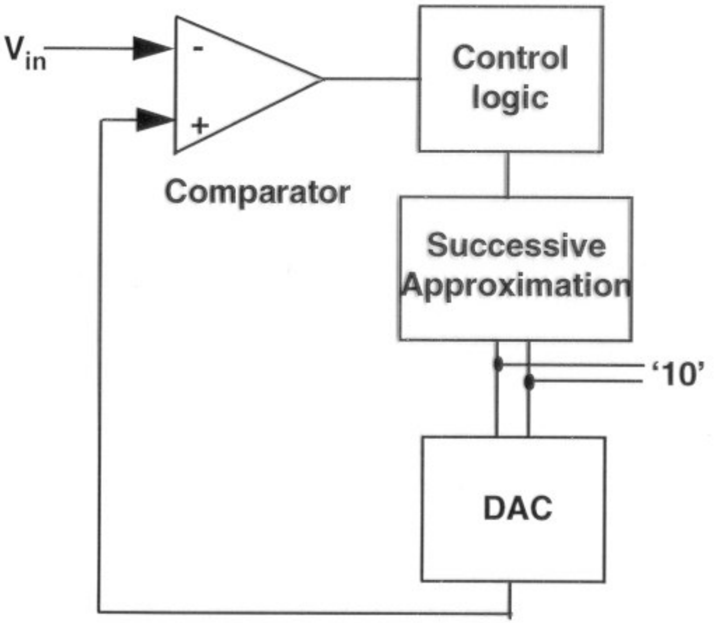
- QUANTIZATION INTRODUCES ERRORS
- INCREASING THE NUMBER OF QUANTIZATION LEVELS IS NOT ALWAYS THE ANSWER

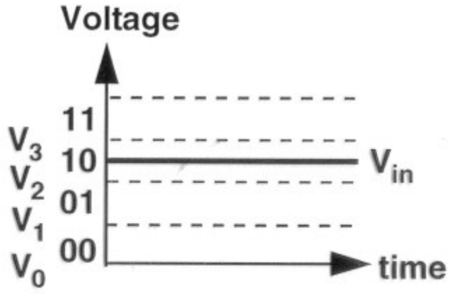


- NON-UNIFORM QUANTIZATION
  - USE MORE LEVELS WHERE THERE ARE MORE VARIATIONS
  - USE FEWER LEVELS WHERE THERE IS NOT MUCH CHANGE

ADC CONVERTERS

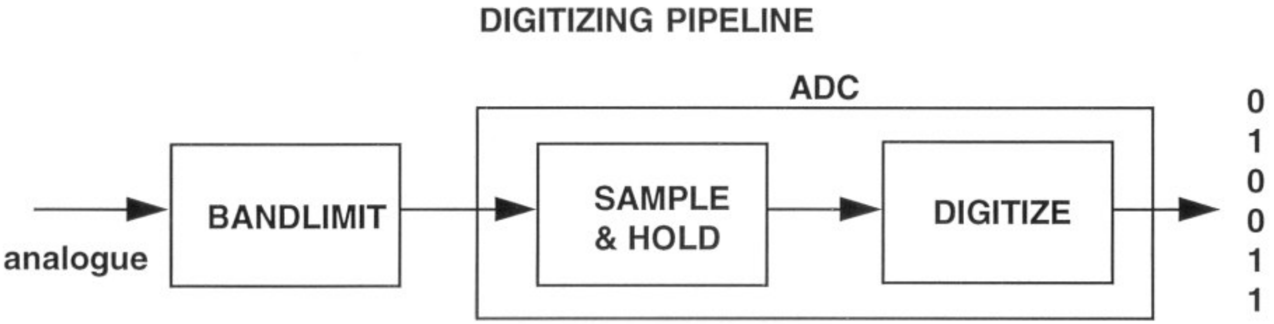
SUCCESSIVE APPROXIMATION ADC





- SET DAC OUTPUT TO V_2 = '01'
- DAC GENERATES ANALOGUE VOLTAGE V2
  $V_{in} > V_2$
- SET MSB TO '1'
- DAC NOW GENERATES V_3
- SET LSB TO '0' SINCE $V_3 > V_{in}$
- DIGITIZE IN TWO CYCLES
- n BITS = n CYCLES

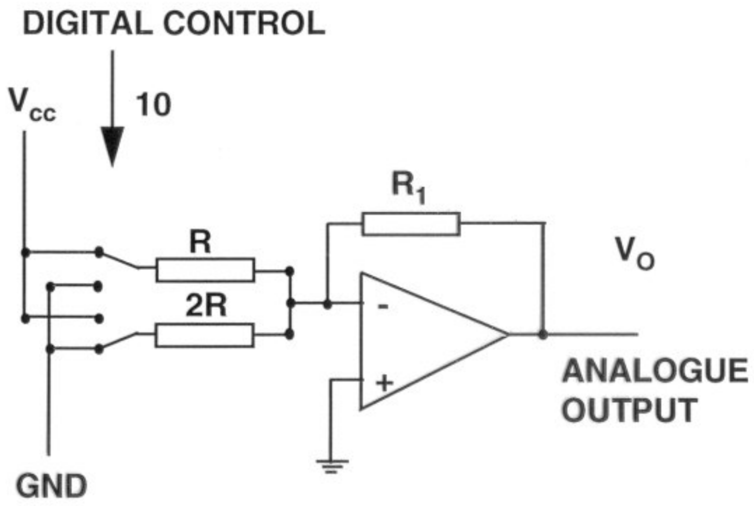
DIFFERENT TYPES OF ADC



- **DUAL SLOPE ADC**
  - SLOW
  - EXPENSIVE
- **FLASH ADC**
  - REQUIRES PRECISION COMPONENTS
- **SIGMA DELTA ADC**
  - USES MOSTLY DIGITAL TECHNOLOGY
  - RELIABLE
  - STABLE

FROM DIGITAL TO ANALOGUE

VOLTAGE SOURCE MULTIPLYING DAC



- FOR '10' DIGITAL CONTROL
  SWITCH IN R TO SUPPLY (V_{cc})
  2R TO GROUND (GND)
  ANALOGUE OUTPUT = (R1/R) * V_{cc}

- GAIN = R1/INPUT RESISTANCE
  THE OUTPUT

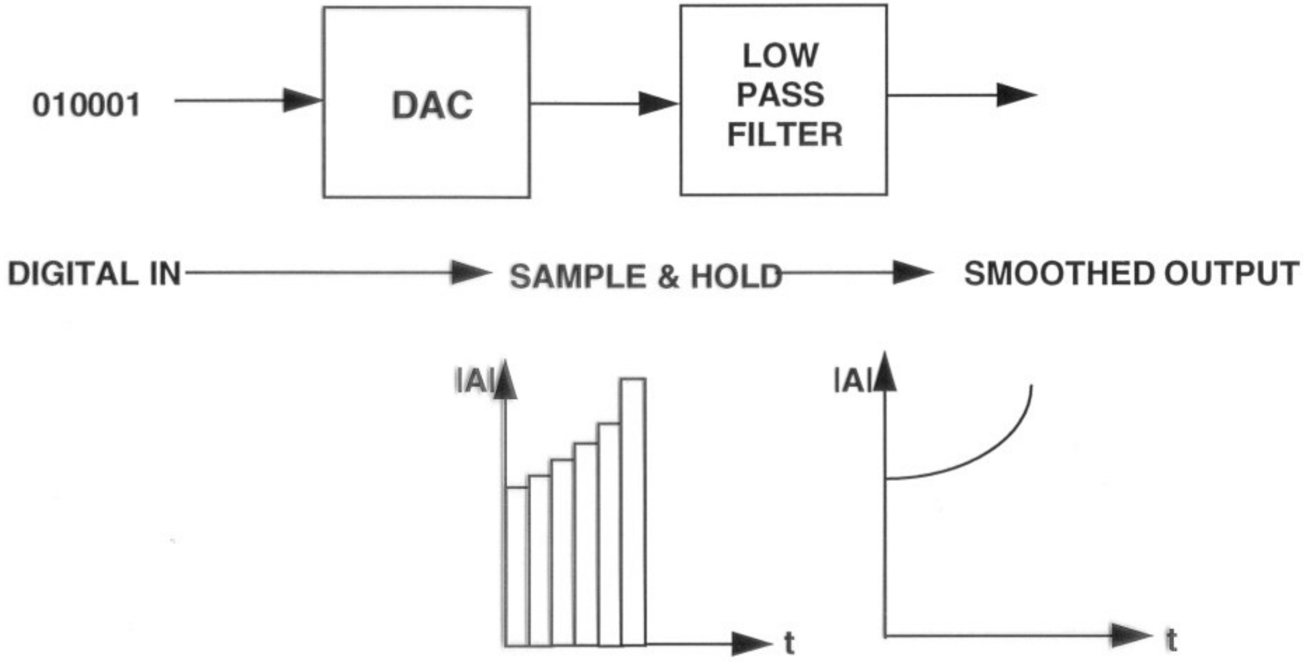
$$V_O = -\left[\ \underbrace{\boxed{V_{in} * (R_1/R)}}_{\text{MSB}} + \underbrace{\boxed{V_{in} * (R_1/2R)}}_{\text{LSB}}\ \right]$$

MSB = MOST SIGNIFICANT BIT
LSB = LEAST SIGNIFICANT BIT

- POSSIBLE INPUTS AND OUTPUTS
  FOR R1 = R

INPUTS	OUTPUTS
11	1.5 V_{cc}
10	V_{cc}
01	0.5 V_{cc}
00	0

SMOOTHING THE OUTPUT

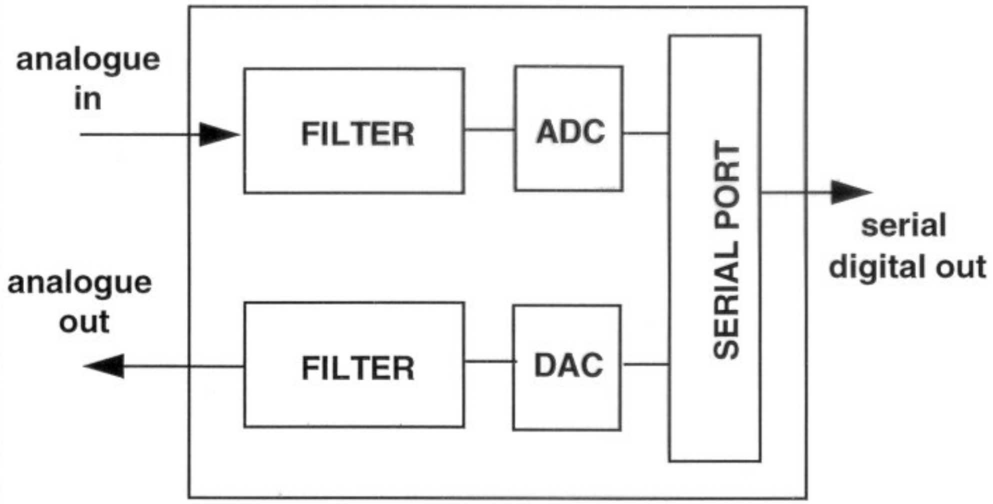


- CONVERT DIGITAL INPUT TO ANALOGUE VALUE
- HOLD UNTIL NEXT DIGITAL INPUT IS CONVERTED
- FINALLY SMOOTH THE OUTPUT SIGNAL

COMMERCIAL CONVERTERS

TLC32040

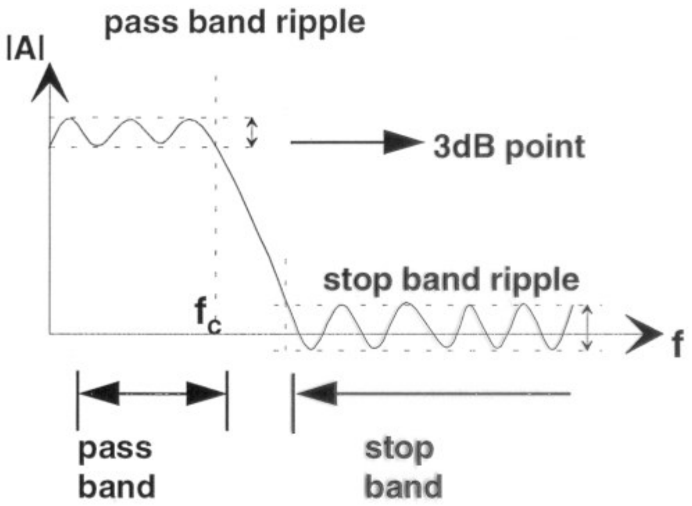
ANALOGUE INTERFACE CIRCUIT (AIC)



- ADC AND DAC ON SAME CHIP
- DIGITIZED DATA SERIALIZED
- INTERFACES TO SERIAL PORT OF DSP
- ANTI-ALIASING FILTER
- SMOOTHING FILTER
- PROGRAMMABLE FILTERS

PERFORMANCE CRITERIA

AMPLITUDE RESPONSE



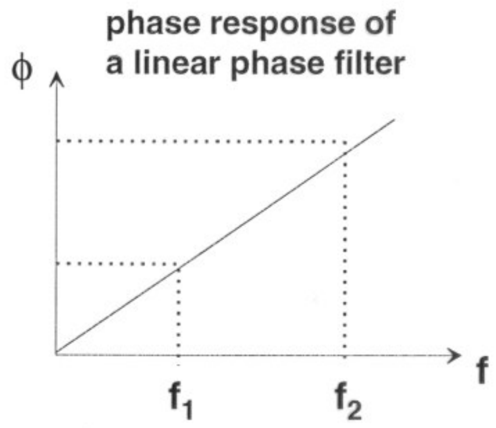
$20 \log_{10} |A|$ = gain in dB

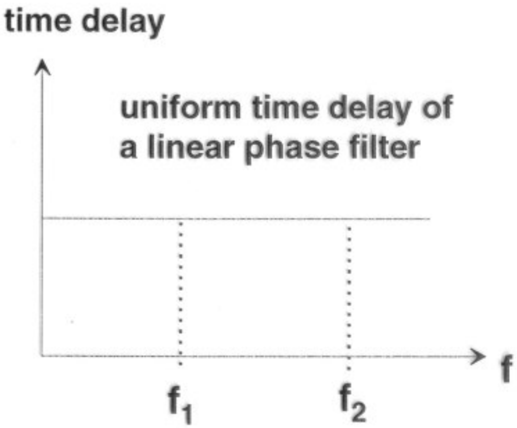
f_c = cut off frequency

GAIN AT 3dB point) (at f_c) = $\dfrac{|A|}{\sqrt{2}}$

- RIPPLE IN PASS BAND CAUSES
  - NON-LINEARITY
- POSSIBLE TO DESIGN WITH NO RIPPLE
- RIPPLE IN STOP BAND IS LESS IMPORTANT
- FALL OFF dB / Decade (Gain in dB / Decade of f)
- STOP BAND ATTENUATES (SAY - 40dB)

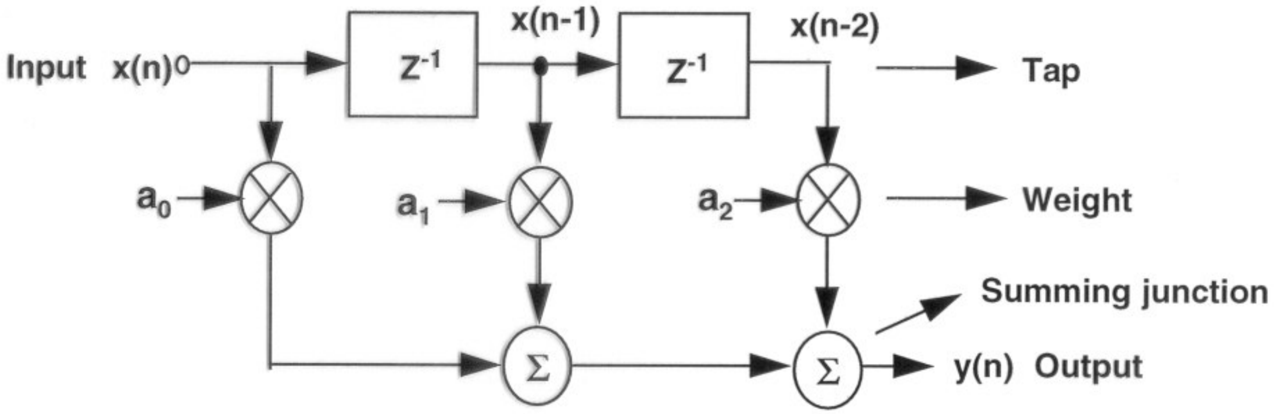
PHASE RESPONSE





- PHASE RESPONSE REPRESENTS TIME DELAY OF DIFFERENT FREQUENCIES
- LINEAR PHASE RESPONSE
  DELAYS ALL FREQUENCIES BY SAME AMOUNT
  TIME DELAYS AT f_1 & f_2 ARE EQUAL
- NON-LINEAR PHASE RESPONSE
  - DELAYS ALL FREQUENCIES BY DIFFERENT AMOUNTS
  - CAUSING DISTORTION TO ORIGINAL SIGNAL
  - IN A MUSIC APPLICATION WE CAN HEAR IT
  - IN A VIDEO APPLICATION WE CAN SEE IT
- LINEAR PHASE IS ONLY IMPORTANT IN PASS BAND
- A LITTLE NON-LINEARITY MAY BE TOLERATED

DIGITAL FILTERS



x(n) sampled analogue waveform, x(0) at t = 0, x(1) at t = t_s, x(2) at t = 2 t_s ...

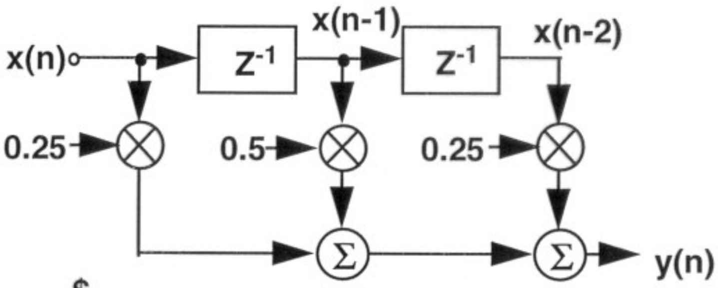
t_s is sampling period, f_s = $\boxed{?}$

a_n = weights (coefficients, scaling factor)

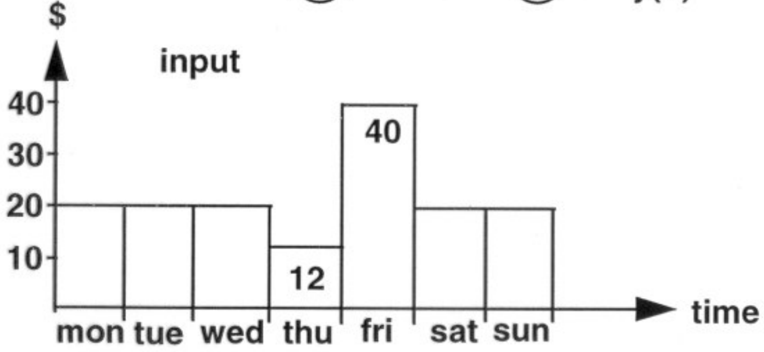
Z^{-1} unit time delay = one sampling period

$y(n) = a_0\, x(n) + a_1\, x(n-1) + a_2\, x(n-2)$

MOVING AVERAGE FILTER



- **ASSUME NO PREVIOUS INPUTS**
  $X(0) = 20;\ X(-1) = 0;\ X(-2) = 0$
- **AND LET**
  $a_0 = 0.25 \quad a_1 = 0.5 \quad a_2 = 0.25$



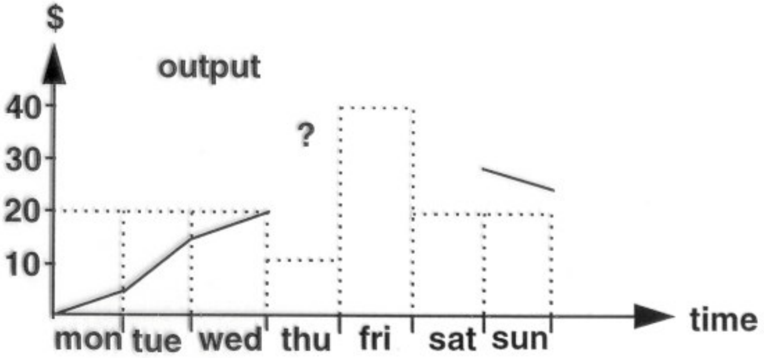
$y(0) = 0.25*x(0) + 0.5*x(-1) + 0.25*x(-2) = 5$

$y(1) = 0.25*20 + 0.5*20 + 0.25*0 = 15$

$y(2) = 0.25*20 + 0.5*20 + 0.25*20 = 20$

$y(3) = 0.25*\boxed{} + 0.5*\boxed{} + 0.25*\boxed{} = \boxed{}$

$y(4) = 0.25*\boxed{} + 0.5*\boxed{} + 0.25*\boxed{} = \boxed{}$



$y(5) = 0.25*20 + 0.5*40 + 0.25*12 = 28$

$y(6) = 0.25*20 + 0.5*20 + 0.25*40 = 25$

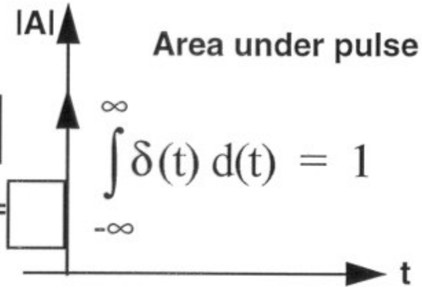
- **MOVING AVERAGE CALCULATION**

WEIGHTED IMPULSE FUNCTION

??

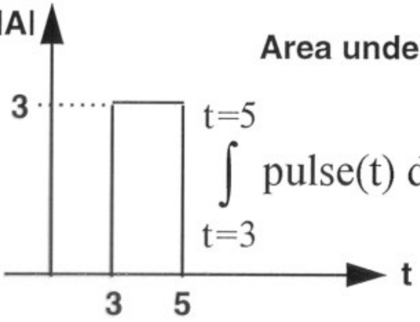
Width = $\boxed{}$

Amplitude = $\boxed{}$



Area under pulse

$$\int_{-\infty}^{\infty} \delta(t)\, d(t) = 1$$



Area under pulse

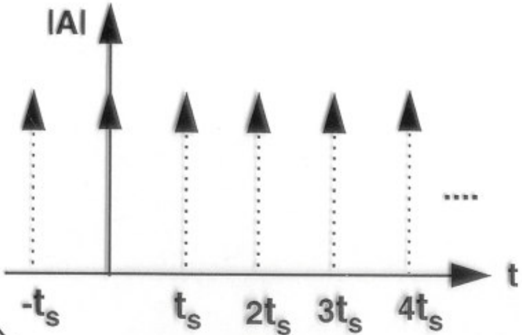
$$\int_{t=3}^{t=5} \text{pulse}(t)\, d(t) = \int_{t=3}^{t=5} 3\, d(t) = 6$$

WEIGHTED IMPULSE FUNCTION

$$\int_{-\infty}^{\infty} A\delta(t)\, d(t) = A \longrightarrow$$

Area = A   Amplitude = ∞

SAMPLING WAVEFORM AS WEIGHTED IMPULSE TRAIN

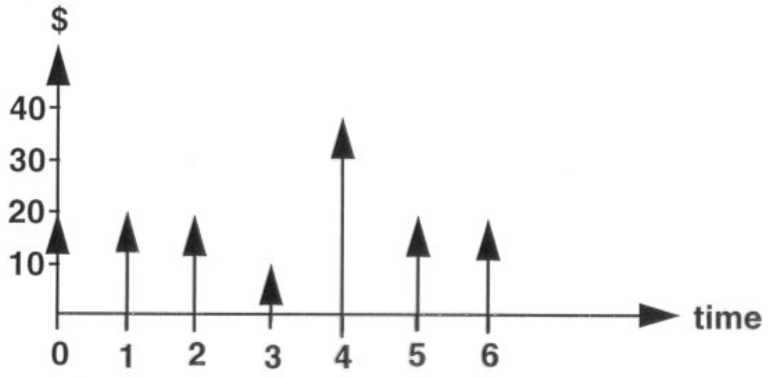


$$\mathbf{s(t)} = \delta(t-\infty) + \ldots\ldots + \delta(t - t_s) + \delta(t) + \delta(t + t_s) + \ldots + \delta(t + \infty)$$

$$\mathbf{s(t)} = \sum_{n=\infty}^{n=-\infty} \delta(t - nt_s)$$

FILTER FUNCTIONS

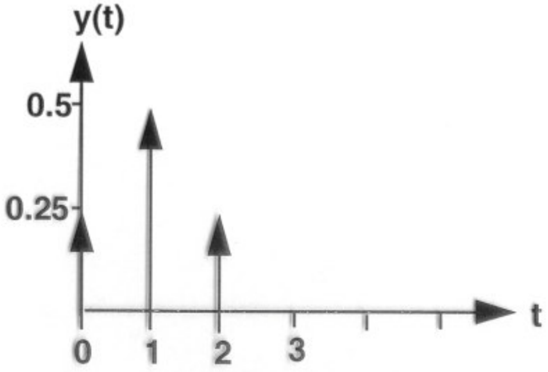
FILTER INPUT AS WEIGHTED IMPULSES



MONDAY'S INPUT VALUE

$$\int_{-\infty}^{\infty} 20\,\delta(t)\,d(t) = 20$$

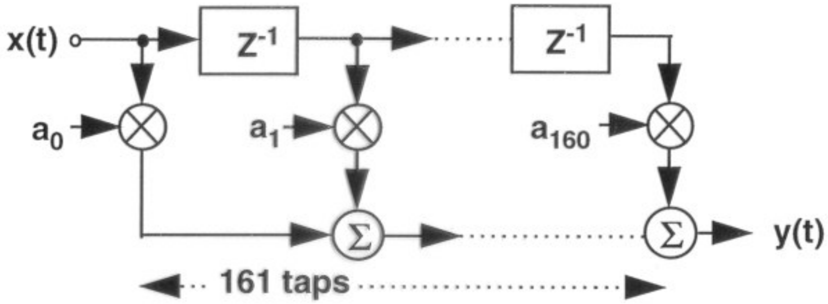
IMPULSE RESPONSE OF FILTER



- OUTPUT WAVEFORM OBTAINED FOR A SINGLE UNITY WEIGHTED IMPULSE APPLIED AT t = 0
- IMPULSE RESPONSE CONSIST OF FINITE NUMBER OF PULSES HENCE FINITE IMPULSE RESPONSE (FIR) FILTER
- IMPULSE RESPONSE MAY BE USED TO OBTAIN RESPONSE TO ANY INPUT

FIR FILTERS

- A FIR FILTER WITH A STEEPER ROLL- OFF

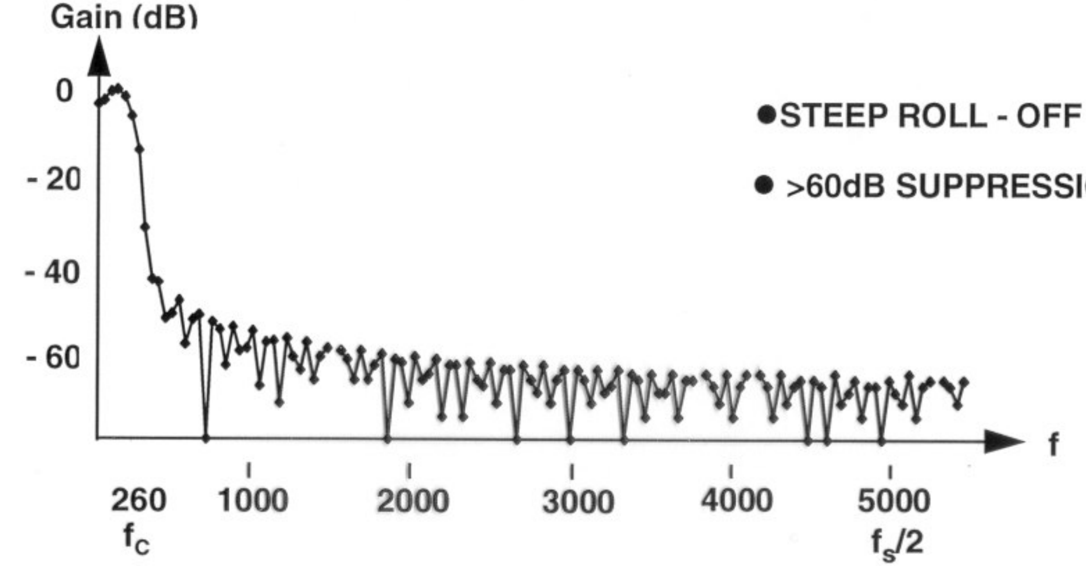


- A MORE REALISTIC FILTER
  DESIGNED USING A SOFTWARE FILTER DESIGN PACKAGE
  - SPECIFICATIONS:
    - CUT - OFF FREQUENCY = 260 Hz
    - STOP BAND ATTENUATION > 60db
    - ROLL - OFF 55dB per 100Hz (VERY SHARP)

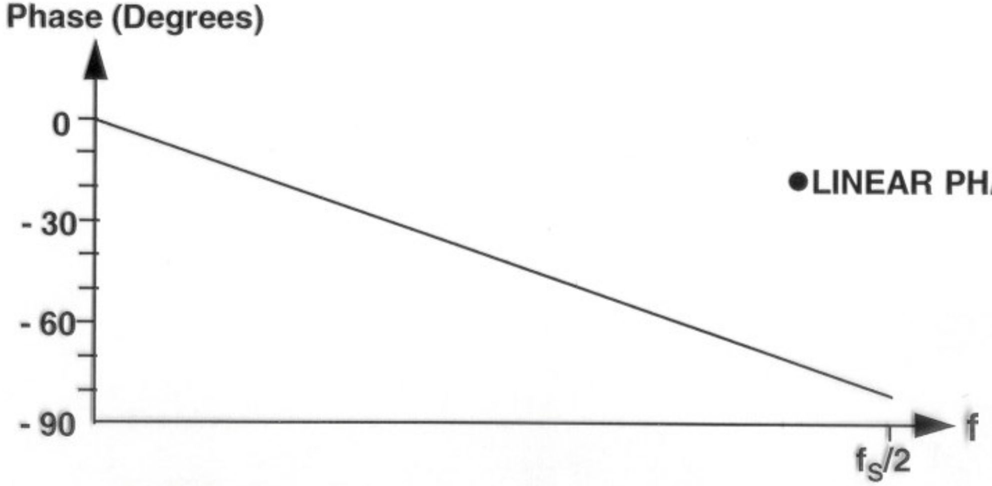
- FILTER WITH 161 TAPS
- 161 DIFFERENT GAIN VALUES

- THIS FILTER IS USED IN OUR DEMONSTRATION

FIR RESPONSE



- STEEP ROLL - OFF
- >60dB SUPPRESSION IN STOP BAND



- LINEAR PHASE RESPONSE

DSP AND DIGITAL FILTERS

ANOTHER TYPE OF DIGITAL FILTER

- INFINITE IMPULSE RESPONSE (IIR) FILTER
  - DELAYS AND WEIGHTS ARE IN FEEDBACK LOOP
  - NON-LINEAR PHASE RESPONSE
  - LESS TAPS, SHARPER FALL OFF
  - MAY BE UNSTABLE

ADVANTAGES OF DIGITAL FILTERS

- PROGRAMMABLE
  CHANGE COEFFICIENTS = NEW FILTER
- POSSIBLE TO IMPLEMENT 'ADAPTIVE' FILTERS
  CHANGE COEFFICIENTS UNDER CERTAIN CONDITIONS
  ON THE FLY

WHY USE DSP FOR DIGITAL FILTER IMPLEMENTATION?
REMEMBER A = B*C + D

$$y(n) = a_0\, x(n) + a_1\, x(n-1) + a_2\, x(n-2)$$

PERFORMANCE ISSUES

NOISE IN DIGITAL FILTERS

- **SIGNAL QUANTIZATION**

  NOISE INTRODUCED IS PROPORTIONAL TO THE NUMBER OF BITS CONVERSION USES

- **COEFFICIENT QUANTIZATION**

  COEFFICIENTS DETERMINE THE BEHAVIOUR OF FILTERS
  MORE SIGNIFICANT IN IIR

- **TRUNCATION**

  0.64 x 0.73 = 0.4672 TRUNCATE TO 0.46
  DOUBLE WIDTH PRODUCT REGISTERS AND ACCUMULATORS HELP REDUCE TRUNCATION ERRORS

- **INTERNAL OVERFLOW**

  0010 + 1111 = 10001 ⟶ SATURATE 1111
  ↳ OVERFLOW

- **DYNAMIC RANGE CONSTRAINTS**

  16 bit ⟶ $20 \log_{10}(2^{16}) = 96\text{dB}$

  32 bit ⟶ $20 \log_{10}(2^{32}) = 192\text{dB}$