Welcome to the New England Radio Discussion Society

AI2Q, May. 2018

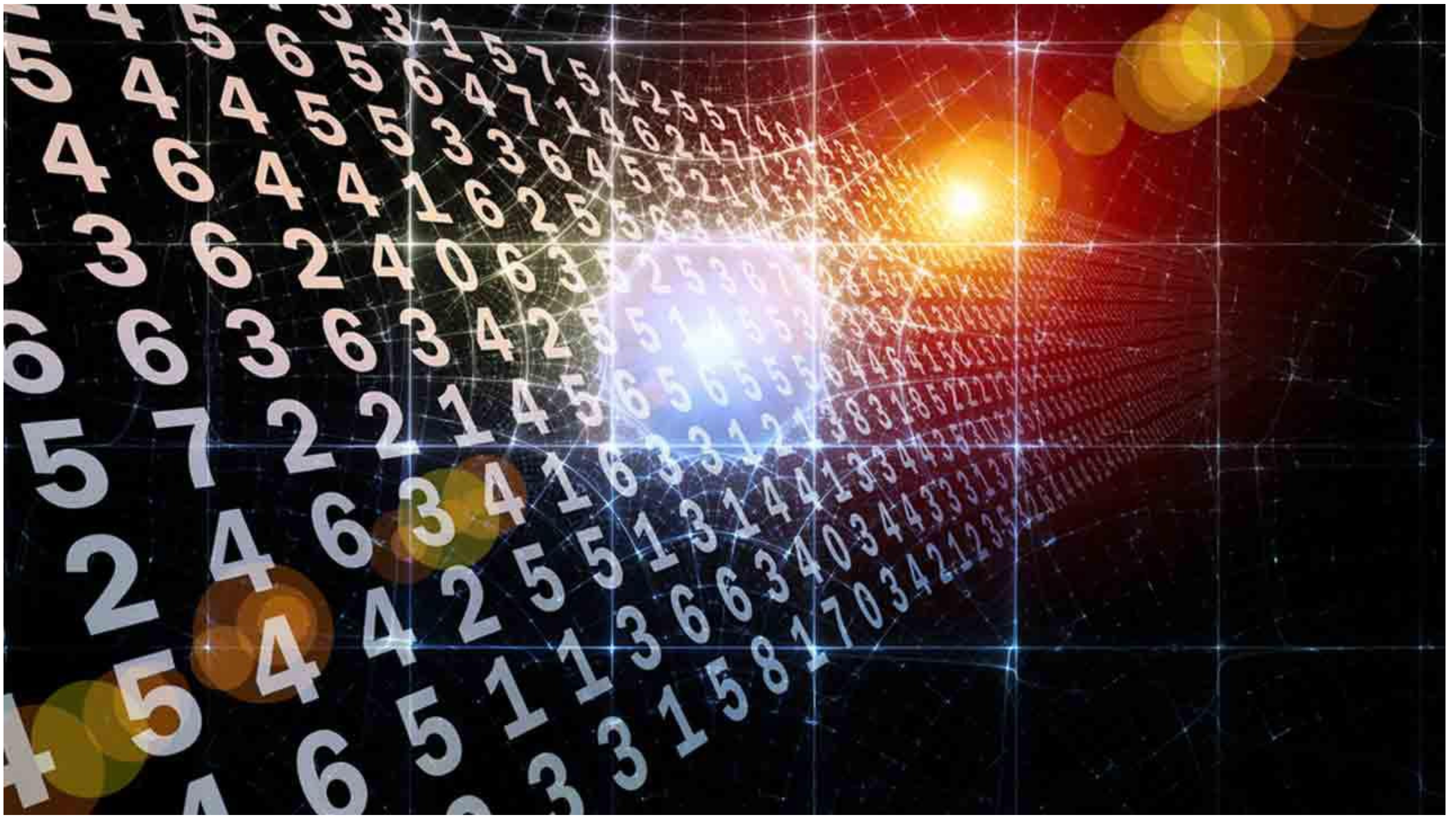
QST ad:

KENWOOD TS-590SG

HF / 50 MHz 100 Watt
Multimode Transceiver
- Enhanced 32 Bit DSP

What’s All This DSP Stuff, Anyway?

What do Digital Signal Processors do?



DSP chips take real-world signals, such as voice, audio, video, temperature, pressure, or position, that have been digitized and then mathematically manipulate them.

DSPs do this in so-called real time. As such, DSPs are said to be deterministic.

A DSP is designed for performing mathematical functions very quickly, such as "add", "subtract", "multiply" and "divide.“ It does this on digital data (0s and 1s).

Ordinary complex instruction set computer (CISC) chips can execute DSP algorithms, but dedicated DSPs use special architectures that can fetch multiple data instructions at the same time.

DSPs are reduced instruction set computer (RISC) chips.

DSP RISC chips are good at manipulating or extracting information from analog signals using few instructions.

DSP chips usually work in conjunction with regular microprocessors or computers.

Again, DSPs like to add, subtract, multiply, and divide.

High-speed RISC DSP chip architectures perform many adds (sums), multiplies, and data transfers – in parallel (concurrently).

DSP algorithms are characterized by these chains of sums and multiplies.

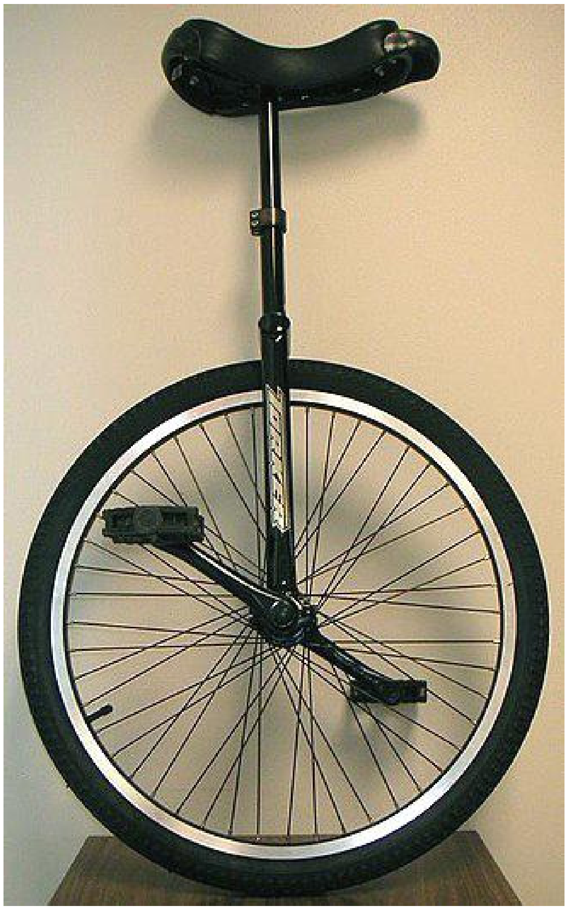
That’s what they like to do, and not many computer instructions are required, compared to CISC microprocessors.

DSP operations do calculations of the form:
$A = B*C + D$

This equation indicates a multiply and an add operation.

The multiply instruction of a general purpose CISC microprocessor is slow compared with a DSP's add instruction. For example, a typical Intel microprocessor uses 10 clock cycles for an "add" and 74 clock cycles for a "multiply."

A DSP can perform the multiply and add operation in just one clock cycle!

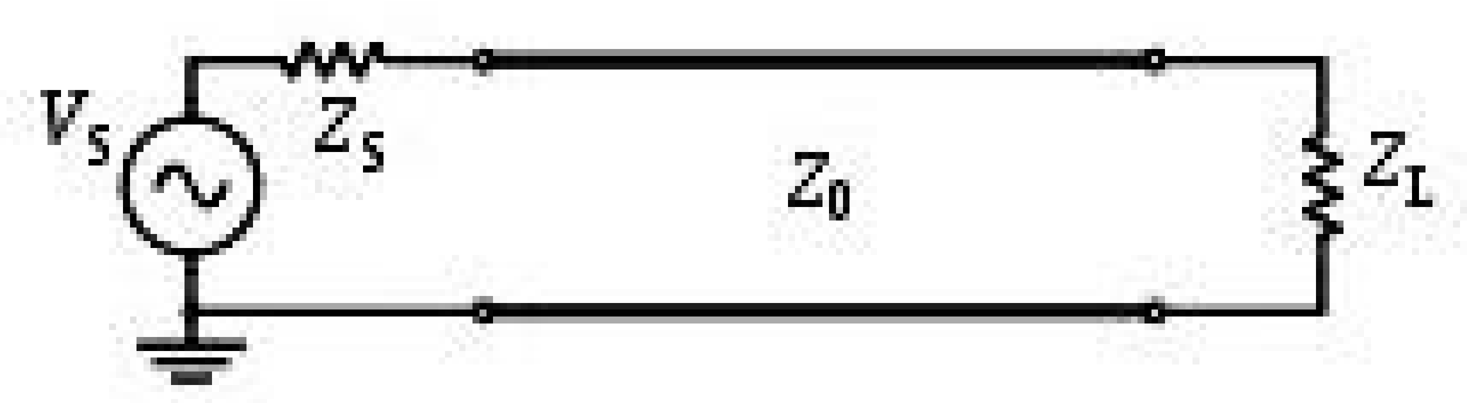


DSPs also pack a specialized instruction that causes them to multiply, add and save the result in a single clock cycle.

This instruction is called a MAC, for Multiply, Add, and Accumulate.

High speed DSPs, clocking at very high frequencies, and processing ever-higher frequency signals, are forcing computer designers to acquire RF skills.

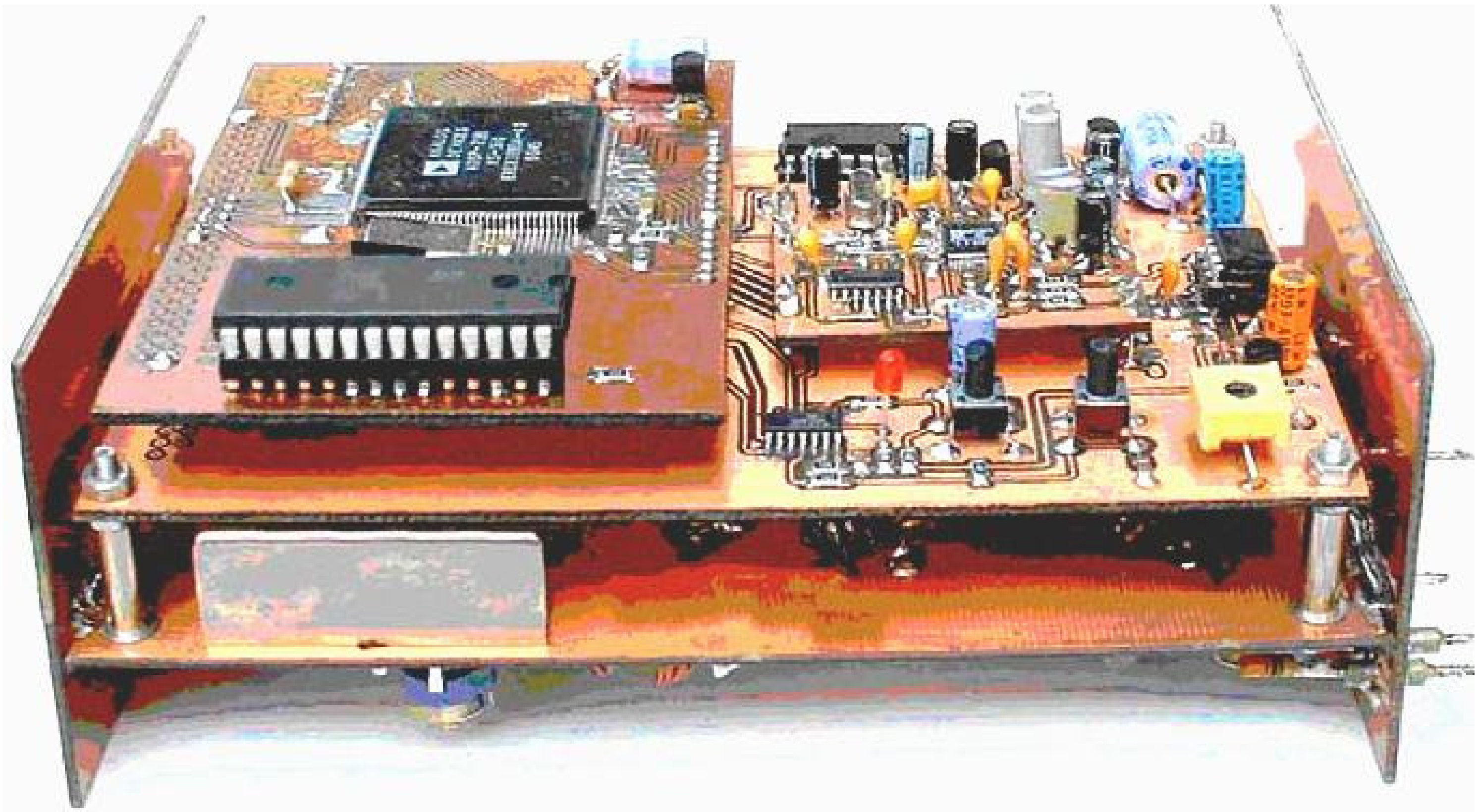
This is especially true when it comes to designing circuit boards, where traces act like transmission lines and antennas.



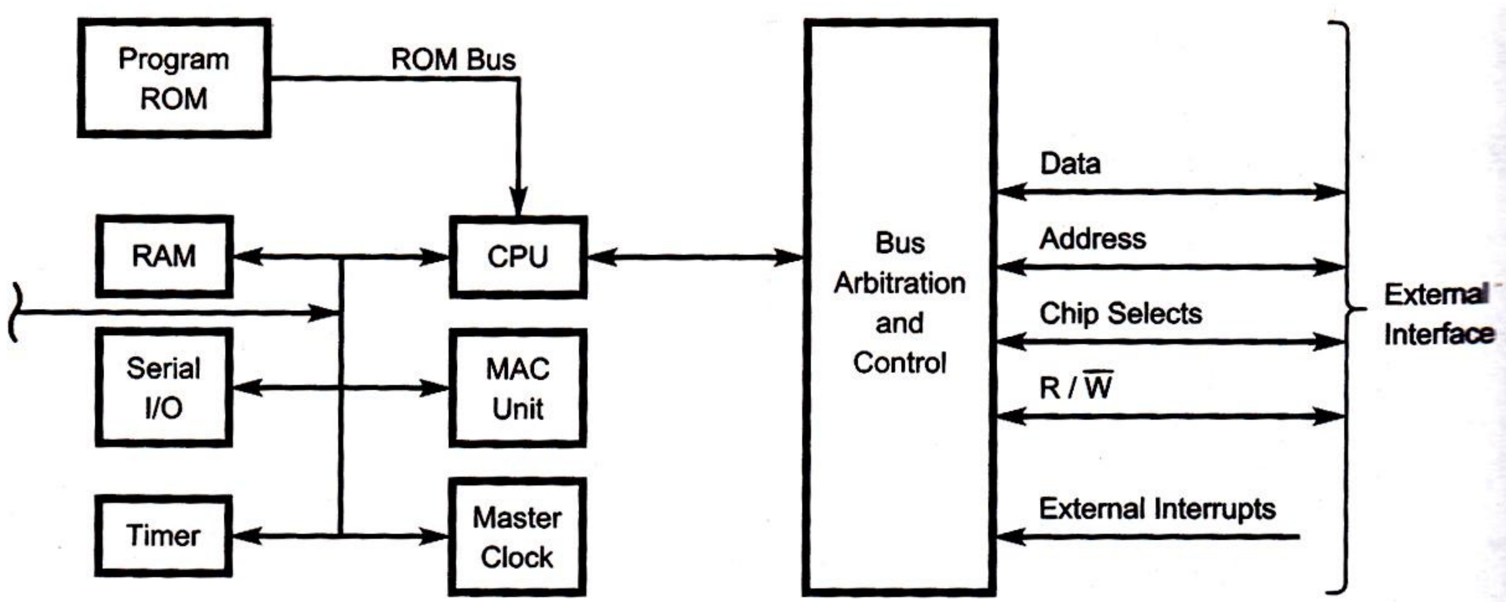
DSPs rely on analog-to-digital converters (referred to as either A/Ds or ADCs) and digital-to-analog converters (DACs). A combined functional block is often referred to as a CODEC (coder-decoder).

System-on-a-chip DSPs reduce the hardware needed for a function, substituting software. SOC DSP also results in price reductions, compared to predecessor all-analog systems and earlier DSPs using separate CODECs.

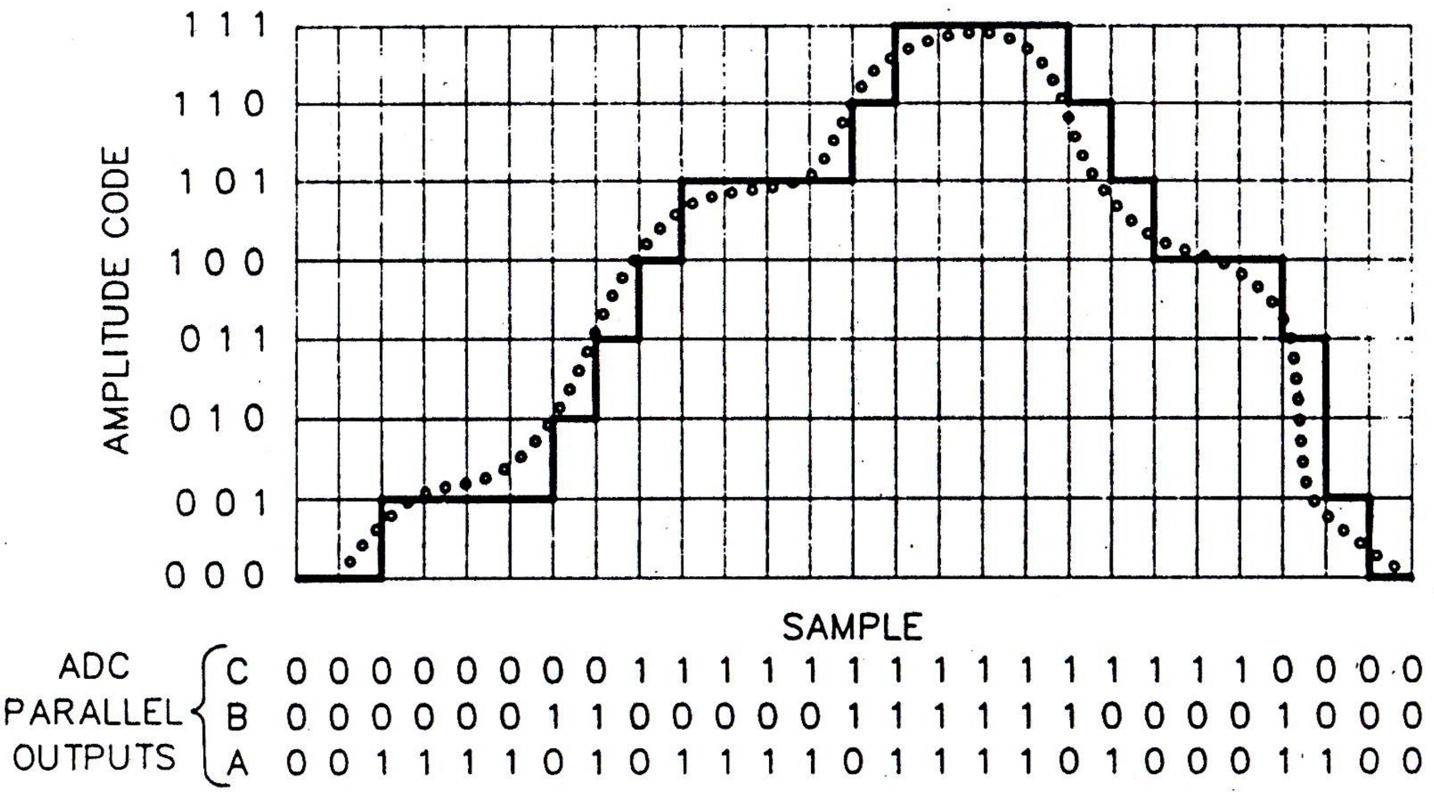
Here’s a board-mounted ADSP2181 DSP chip from Analog Devices Inc. and its mating AD1881 CODEC.



Here's a typical DSP internal hardware block diagram



A/D converters transform analog signals into sequences of digital numbers.

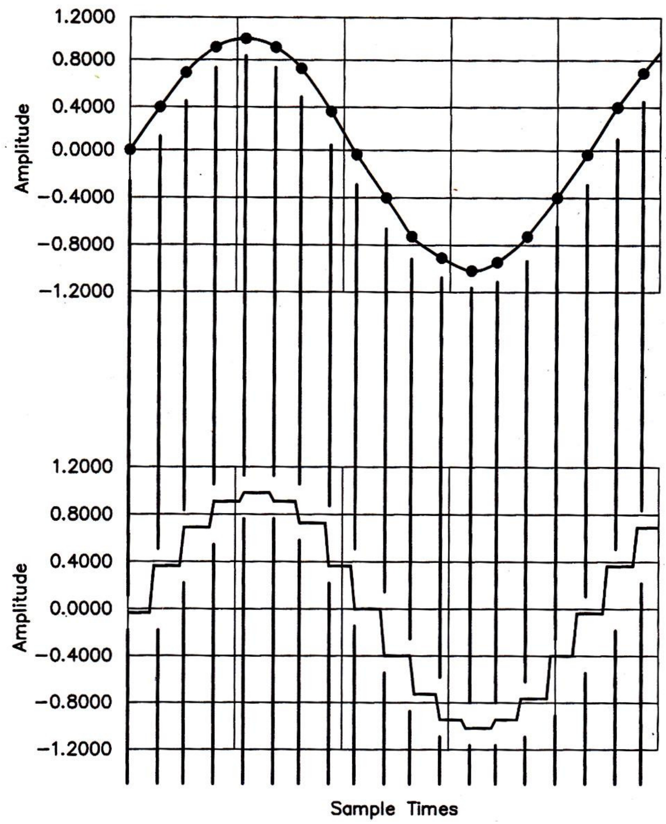


Sampling a signal is the process of making periodic measurements and storing the result (the samples).

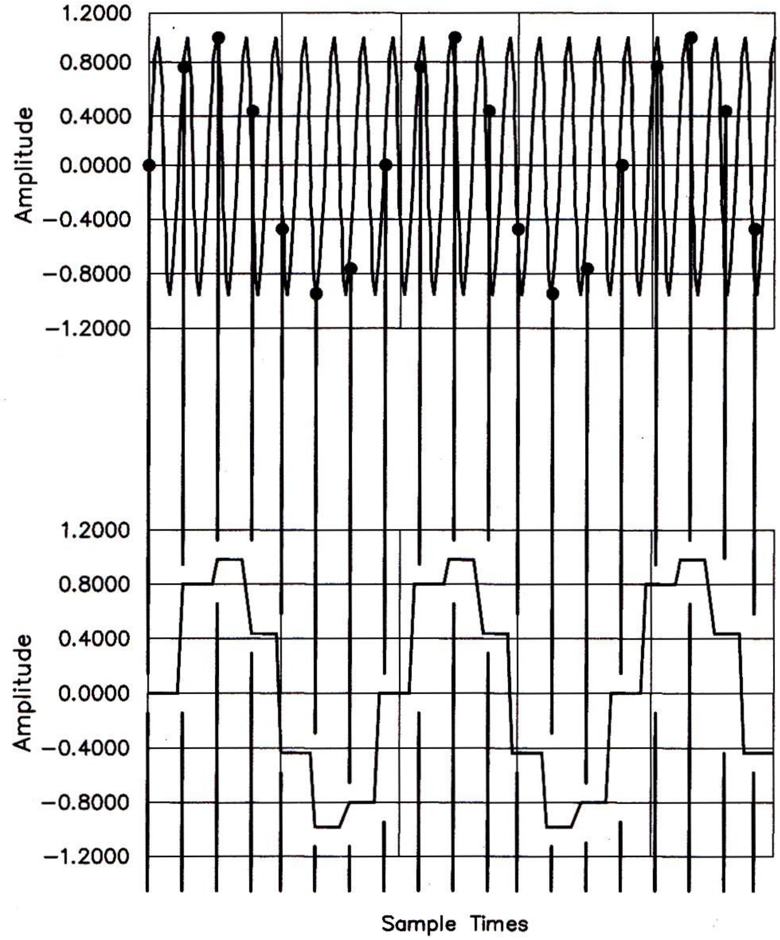
It is not necessary to capture <u>all</u> of the information in a signal.

Here's a sampling of a sine wave of a frequency much lower than the sampling frequency.
→

Neat!



Here’s an example of sampling of a sine wave that’s greater than the sampling frequency. →



Beat!

Sampling Theorem

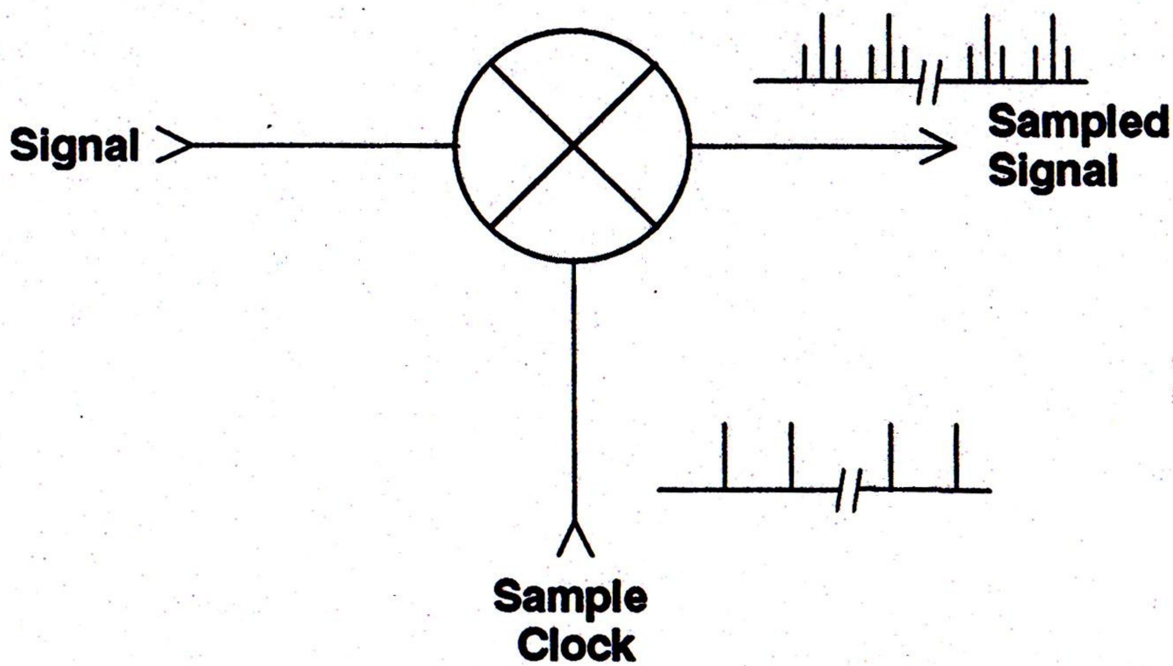
All input signals must be at frequencies less than half the sampling frequency.

$$0 \leq f < F/2$$

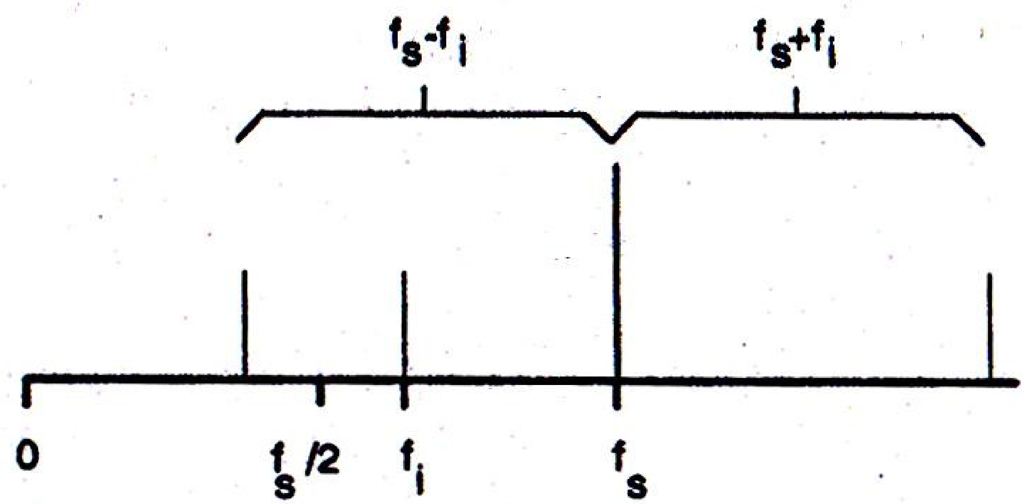
Where F is the sampling frequency, and f is the input frequency.

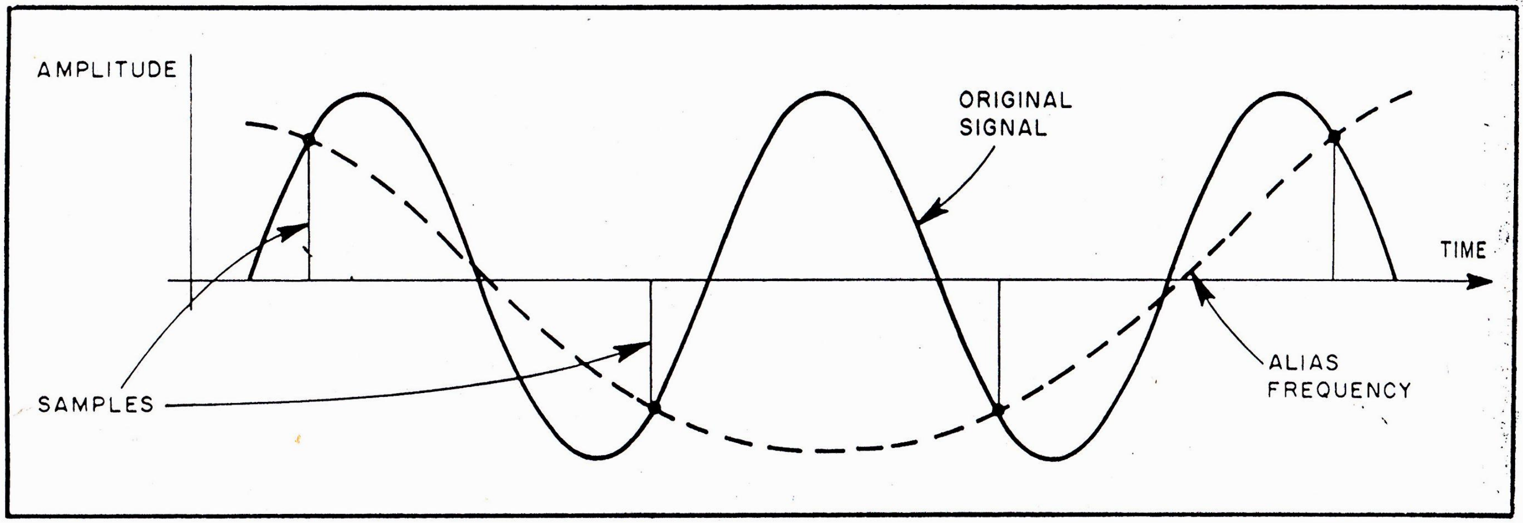
The minimum sampling frequency needed to properly sample the input signal is twice the highest frequency of the signal. This sampling frequency is called the Nyquist rate.

Once digitized in an A/D an effect called aliasing creates sum and difference frequencies, and these new aliases may not be desirable.



The aliases are the result of sums and differences between the desired digitized signal f_i and the sample clock f_s .

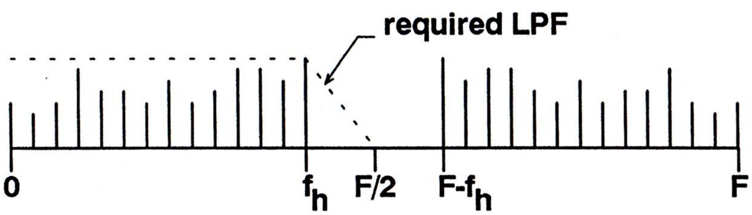




Note how aliasing occurs: the sampled values could have come from either the original signal or a signal at the alias frequency. Once the signal is digitized, it consists only of the samples, and there is no way to determine which of the possible signals was the original one.

Low-pass filters at the input to the A/D can provide anti-aliasing.

Must sufficiently reject all signals above F/2 to eliminate aliasing. The closer F/2 is to the desired highest input frequency, the more complex the filter must be.



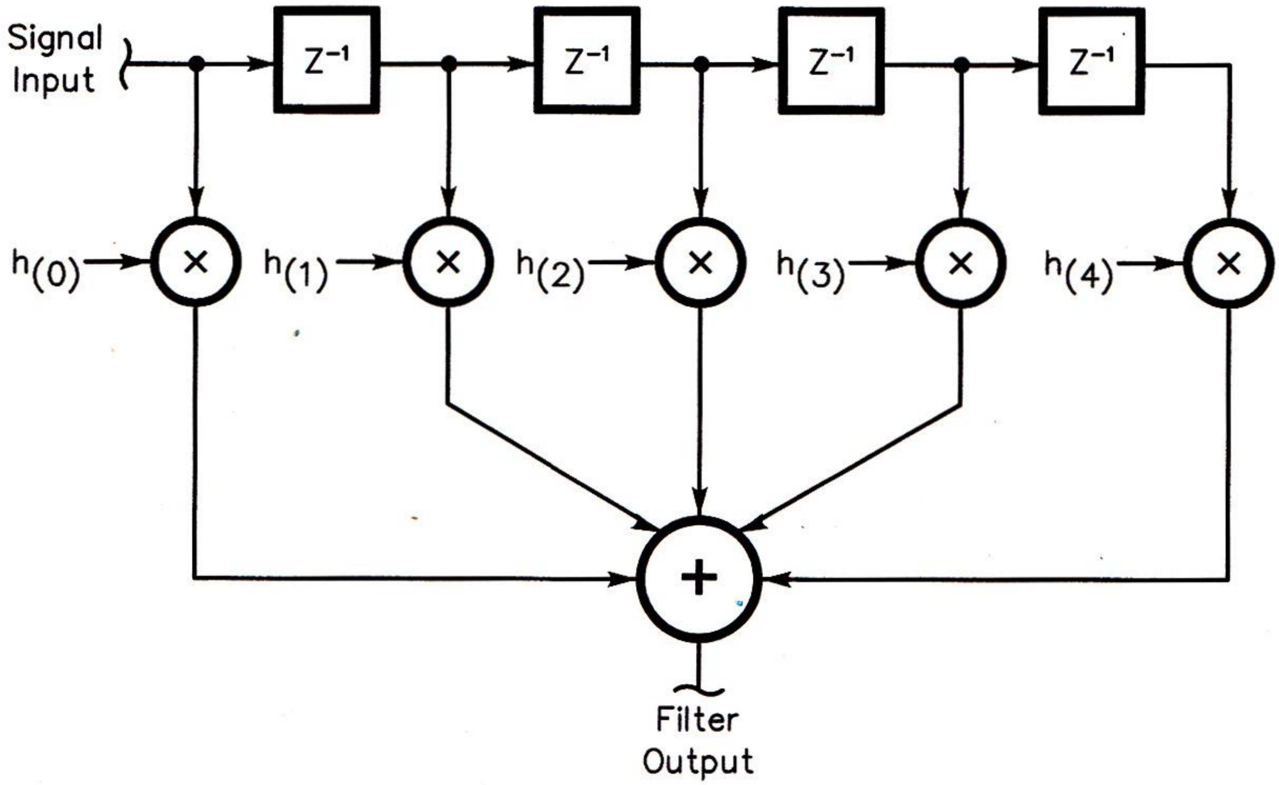
f_h= highest input frequency, F = sampling frequency]

Outputs from a DSP system's DACs also frequently demand sampling filtering.

In any case, the DSP quickly crunches the numbers!

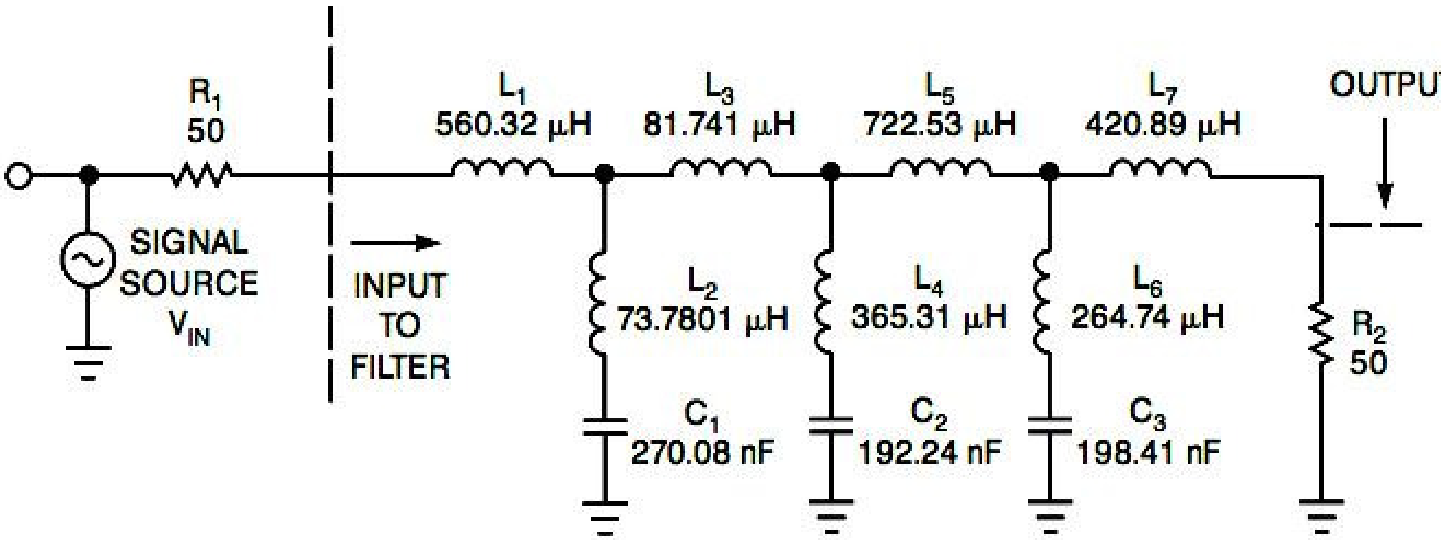
CORRELATION COMPARES	$R_N = \sum_{M=0}^{N-1} X(M)\, Y(M+N)$	CROSS-CORRELATION
CONVOLUTION FILTERS	$Y_N = \sum_{M=0}^{N-1} H(M)\, X(N-M)$	"FINITE IMPULSE RESPONSE" FILTER
TRANSFORMATION FINDS FREQUENCY CONTENT	$F_K = \sum_{M=0}^{N-1} F(N)\, EXP^{-J2\pi MK/N}$	DISCRETE FOURIER TRANSFORM

This filter block diagram depicts the multiplies and accumulates sequence.

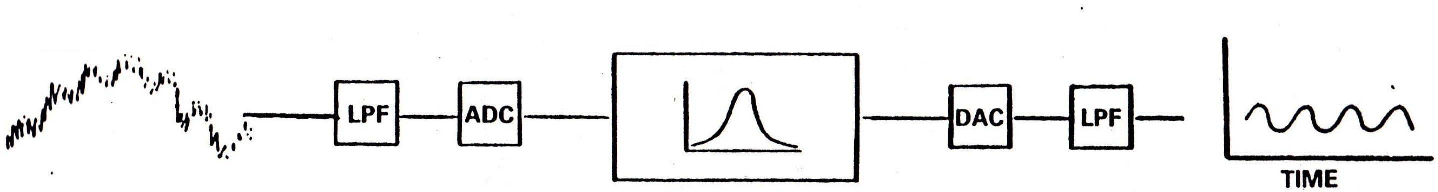


By comparison, here's what a precision discrete-component analog filter might look like. This example filter design uses "oddball" component values. These parts are also subject to aging over time.

A mathematically-defined DSP filter eliminates those problems.

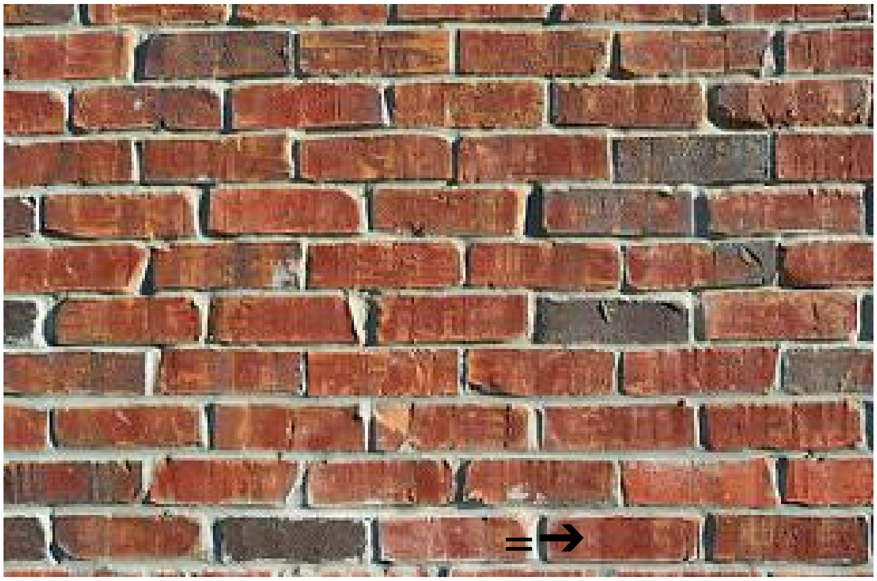


Filtering is a common DSP application in Amateur Radio rigs. Good filtering can separate desired signals from nearby QRM. A DSP can act as a very effective filter.



DSP filters and mixers are also used in transmitter circuits.

Mathematical DSP filters can provide so-called brick wall filtering.



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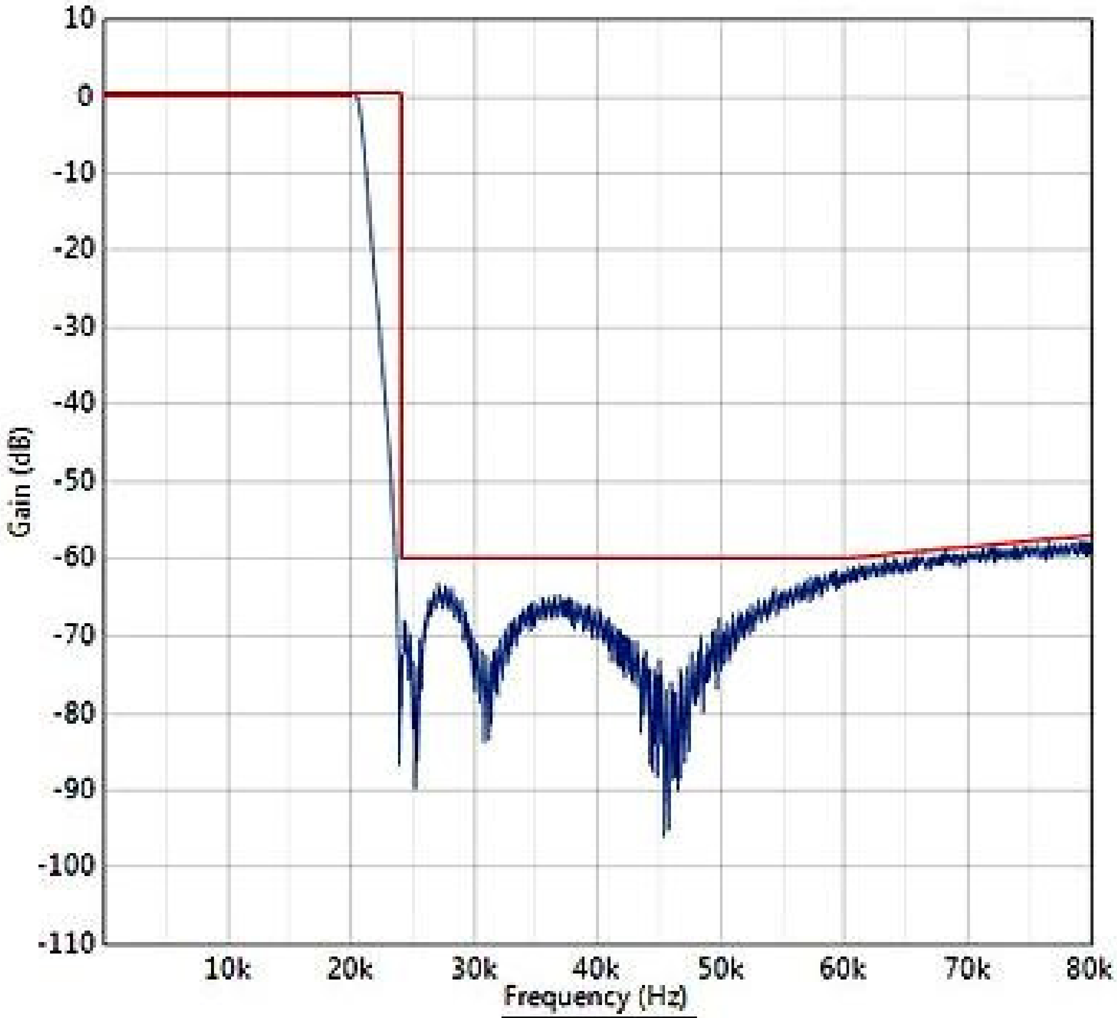
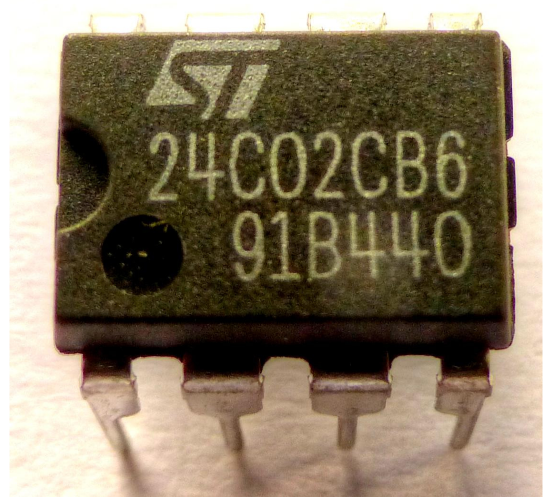


10
0
-10
-20
-30
-40
-50
-60
-70
-80
-90
-100
-110
Gain (dB)
10k
20k
30k
40k
50k
60k
70k
80k
Frequency (Hz)

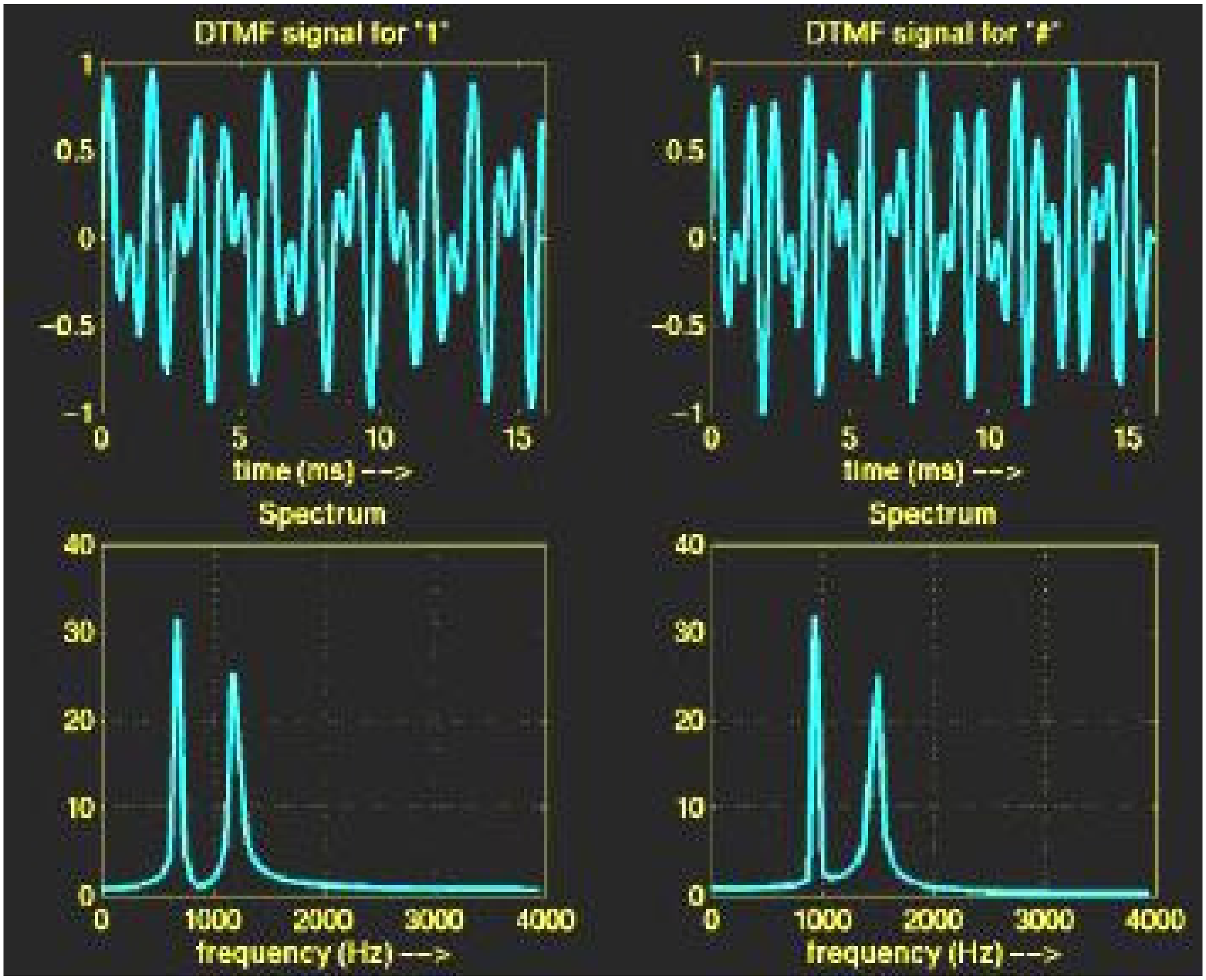
In addition to filtering (LPF, HPF, and band-pass filters) DSPs can also be used to mathematically implement coding for modulation, de-modulation, spectrum analysis and display, imaging and spatial pixel filtering, data compression, direct digital synthesis (DDS) for frequency generation, spectrum analysis, VOX circuits, noise reduction, RADAR, and speech recognition, to name just a few applications.



The numerical-manipulation code for DSP often resides in non-volatile memory, such as an EPROM chip.

Many DSPs include program memory on-chip, as well as RAM. Programming determines function!

This example shows how DSP-based spectral analysis can “de-code” a DTMF TouchTone^{tm} signal.



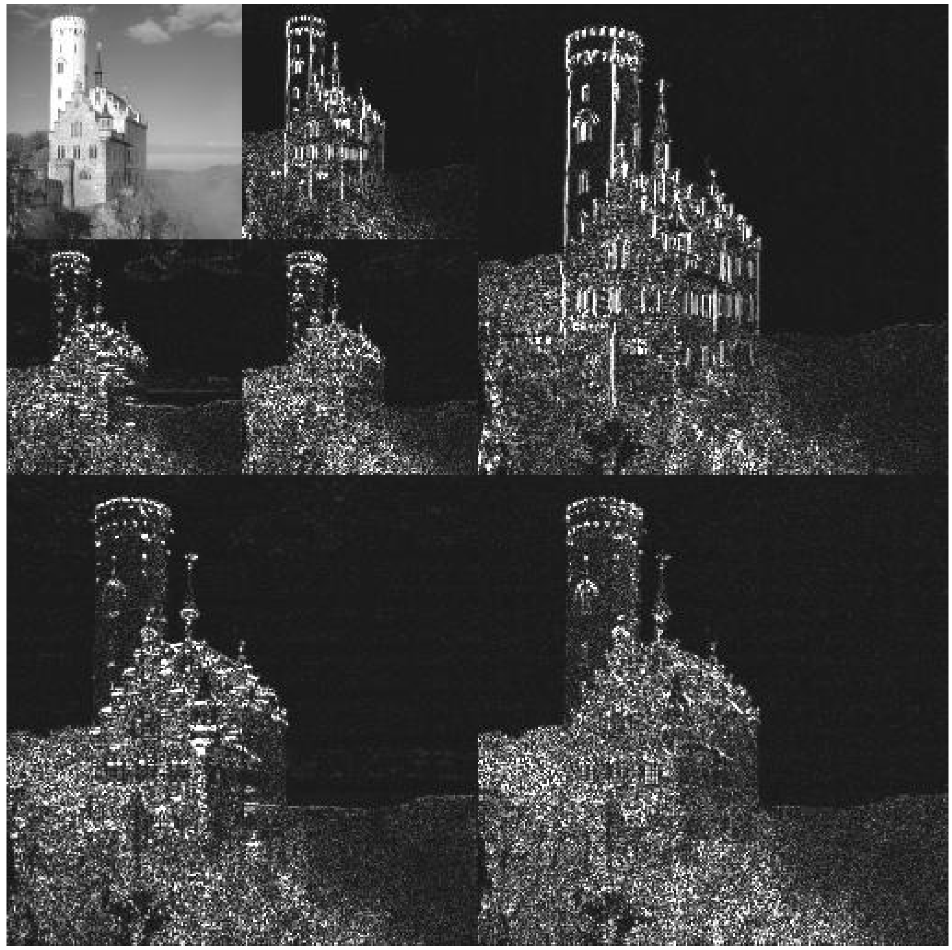
DSP systems -

- Don’t drift with temperature or power supply fluctuations

- Don’t exhibit long-term component aging

- Provide identical performance from rig to rig, without need for individual tuning or adjustments

Here's a 2D transform used to create a JPG file. The original image is high-pass filtered, yielding the three large images, each describing local changes in brightness in the original. It's then low-pass filtered and down-scaled, yielding an approximation; this is high-pass filtered to produce the three smaller detail images, and then low-pass filtered to produce the final image.



DSPs are used for facial recognition processing

Have fun!

Vy 73, AI2Q