

# Autowah: a dynamic audio filter on a TMS320VC5510 DSK

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**W**ith the WAH effect a musician can change the timber of the sound moving a pedal. The AUTOWAH can properly filter the signal, with no human action. An envelope filter figures the frequency to enhance it and send it to a bandpass. It enhances the right frequencies at the right time. It's a simple smart filter.

The envelope filter is a rectifier and an  $RC$  lowpass. It is designed in continuous time and then digitalized: so that only one parameter has to be set,  $RC = 0.3$ , instead of two.  $RC$  set filter's delay. Here is the difference equation:

$$y_n = b \cdot |x_n| + b \cdot |x_{n-1}| - a \cdot y_{n-1} \quad (1)$$

where

$$b = \frac{T}{2 \cdot RC + T} \quad a = \frac{2 \cdot RC - T}{2 \cdot RC + T}. \quad (2)$$

NB:  $a$  and  $b$  are found with Tustin's algorithm.

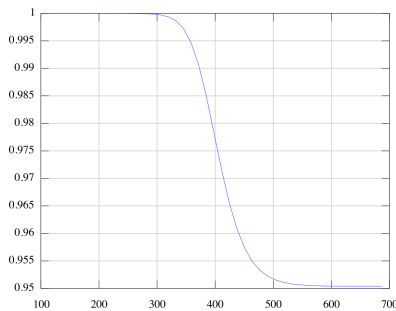
For the bandpass, a simple  $II^o$  IIR filter centered in  $\Omega_0$  is used. Here is the difference equation:

$$y_n = A^2 \cdot x_n - A^2 \cdot x_{n-2} + 2A \cdot c \cdot y_{n-1} - A^2 \cdot y_{n-2} \quad (3)$$

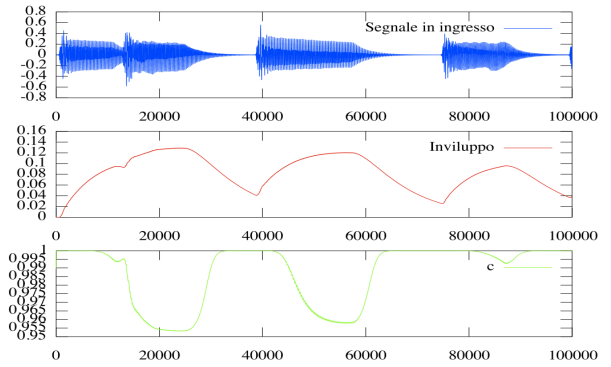
where  $A = 0.9$ . The parameter that connects the units of the filter is

$$c = \cos(\Omega_0) = \cos(2\pi \cdot \frac{f_0}{f_s}) \quad (4)$$

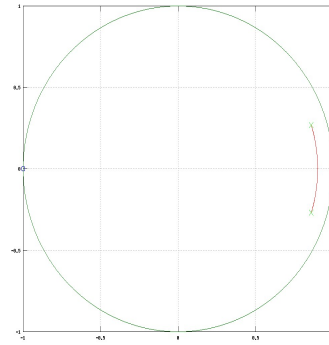
where  $f_s$  is the sample rate and  $f_0$  a  $\tanh$ -shaped function of the envelope. As the DSP is good for multiplying and accumulating, it is better to give to it pre-computed values in array instead of having the  $\tanh$  calculated by the DSP. A 39-long array is used.



Look-up table:  $c$  versus *envelope*



Example of *input*, *envelope* and  $c$  versus *sample*



Poles of the bandpass while  $c$  decreasing

While I was debugging my code I needed to use the LUT periodically just to try its output values with no input: I left this part in the main code because it sounded good. Several distortions came out too, so I included a strong preamplifier in the end of my work. These effects together make a nice bass synth.

A variable *corr* is used to correct the output volume and is updated through the program when different effects are switched on.

The header *CONTROLLI.H* contains four parameters that can be changed through a watch window and three bool variables, connected to the DSK switches.

Also read *main.c*, *lut.h* and *filtro.h* for more details and explanations.

The whole work has been done on the TMS320VC5510 DSK board using Code Composer Studio 3.1 in IPL2, Image Processing Lab, DI<sup>3</sup> (former DEEI), University of Trieste.