Non-Linear Transformations

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1 Non-linear effects

In signal processing, a nonlinear filter is a filter whose output is not a linear function of its input.

Both continuous-domain and discrete-domain filters may be nonlinear. Nonlinear filters have many applications, especially in the removal of certain types of noise that are not additive. As an example, all radio receivers use non-linear filters to convert **kilo** to **gigahertz** signals to the audio frequency range, and all digital signal processing depends on non-linear filters (ADC) to transform analog signals to binary presentation.

Moreover, Linear filters are often used to remove noise and distortion that was created by nonlinear processes, simply because the proper non-linear filter would be too hard to design and construct.

1.1 Problem

Clipping a special case of distortion, so lets describe some of it's properties:
The effect is based on the property of both tube and transistor amplifiers to introduce nonlinear distortions into the signal, especially if it is close to the maximum possible for a particular amplifier. Overload (especially a multi-stage tube amplifier) differs from simple clipping in that the output signal has a complex dependence of the spectral components on the amplitude and spectral composition of the input signal, in contrast to the elementary limiter. Traditionally, the sound of overdriving a tube amplifier is valued above the sound of overdriving a transistor amplifier.

```
function f = CLIP_F(x, l)
    f = x
    f(abs(x) > l) = sign(f(abs(x) > l)) .* l
endfunction
```

Figure 1: Clipping effect

1.2 Solution

1.3 Results

With an increase in the amplitude of the output signal, the coefficient of nonlinear distortion increases. The non-linearity of the characteristics of the amplifier depends on many factors

Let's compare results of clipping, left side it is original signal one, right - after clipping:

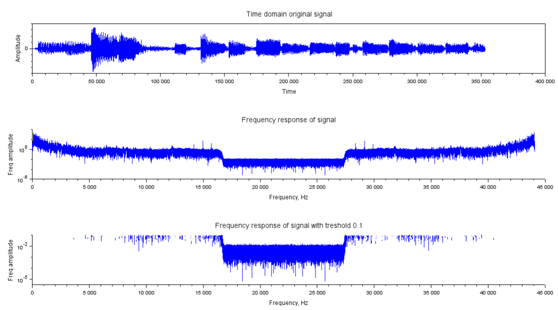

Figure 2: Original

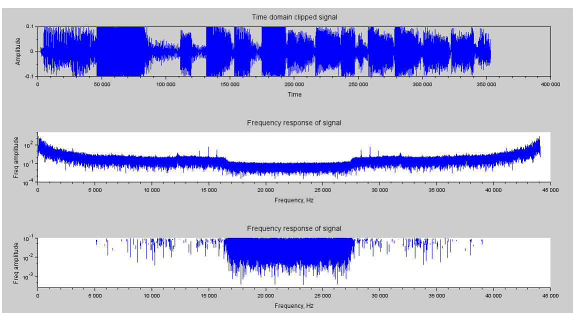

Figure 3: Clipped

Here we see difference if we will pay attention only on frequency domain part where signal ¡ our threshold. Also in time domain part we see that we clipped our signal on amplitude limit.

1.4 Problem

Here we see difference with clipping, first it is amplitude level, second overload signal at threshold level looks smoother than original and clipped. Also the extra harmonics appeared in the frequency part. As in the case of clipping, these harmonics correspond to integer multiples of the original frequency.

1.5 Solution

```
function f = DIST_F(x, a, b)
    f = x
    for i = 1:length(f)
        f(i) = a * atan(b*f(i))
    end
endfunction
```

Figure 4: Distortion effect

1.6 Results

Here results of distortion effect:

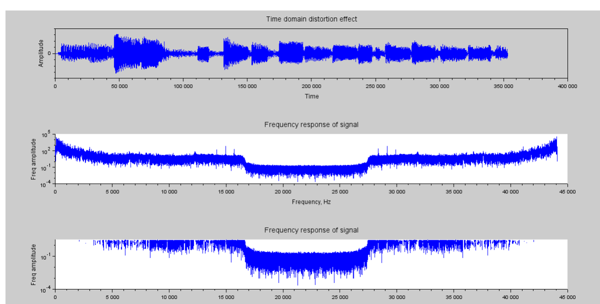

Figure 5: Distortion

1.7 Bonus

Simple observation we can apply to distortion effect, if we will look at original atan function:
We see that it is non-linear relation outputs from inputs.
For clipping effect we also can prove non-linearity by definition like:
$sign(x + y) * a \neq sign(x) * a + sign(y) * a$ because if one of the elements bigger and have opposite sign than another, so results will be not equal.

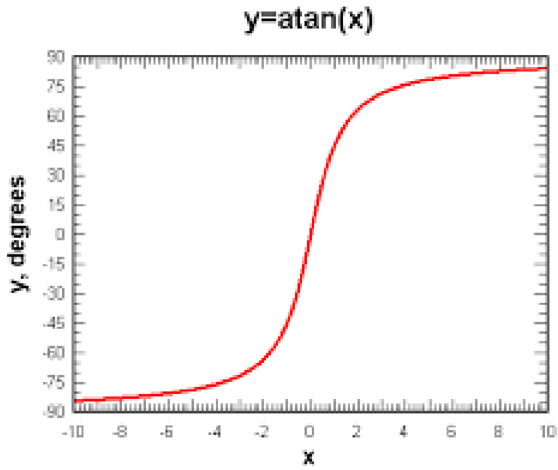

Figure 6: Caption

2 References

Git hub repo with source code and signals
Theory about Overloading in signals