Gautebøye - Instrument Response

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Introduction

Here is presented the theoretical derivation and analysis of the analog instrument response of the digitizer. The digital filtering during AD conversion is briefly discussed, but its properties are well defined in the manual of the ADC[6].

1 Hydrophone decoupling

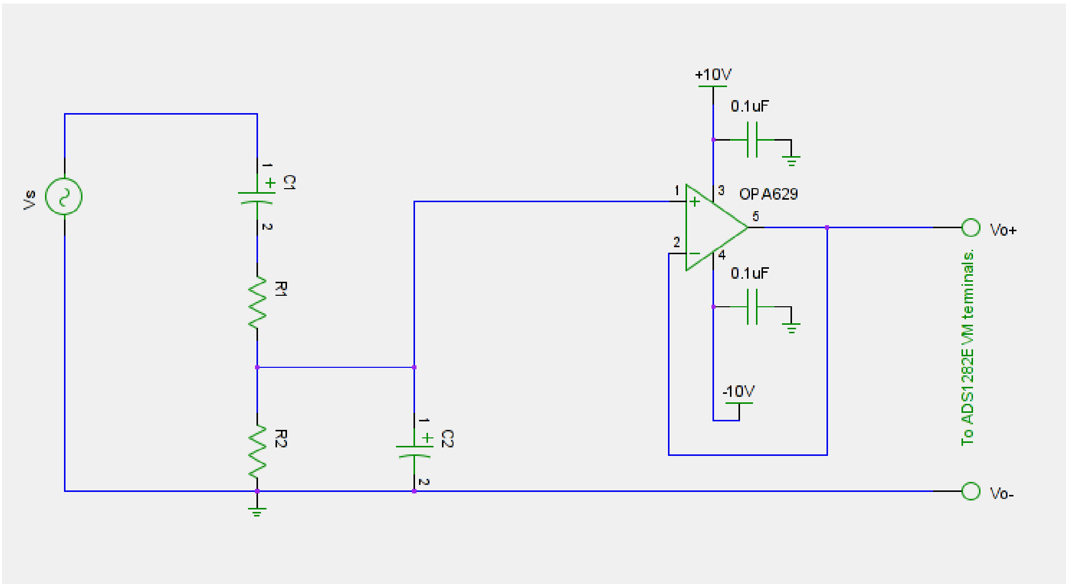

Figure 1: Hydrophone decoupling circuit

The figure above (1) shows an equivalent circuit of the deployed input decoupling to the hydrophone.

$$C_1 = 47\mu F$$
$$C_2 = 330pF$$
$$R_1 = 58.33k\Omega$$
$$R_2 = 41.67k\Omega$$

The circuit was based on the suggested decoupling in figure 2 seen below, it was designed to have the same properties, but also low-pass the signal at cutoff frequency f_{LP_1}. The original design already decouples, and high-passes, the signal at frequency f_{HP}.

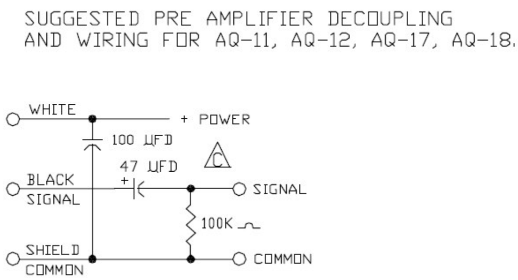

Figure 2: AQ-18 suggested decoupling and wiring [1].

2 Derivation of transfer function

i_1 and i_2 signify the current loops passing through respectively left and right loop. Using Kirchoff's equation:

$$V_s = i_1(Z_{C_1} + Z_{R_1}) + (i_1 - i_2) \cdot Z_{R_2} \quad (1)$$
$$0 = (i_2 - i_1) \cdot Z_{R_2} + i_2 \cdot Z_{C_2} \quad (2)$$

V_s is the signal source, the hydrophone. See separate data sheet for hydrophone frequency response. V_o, output, is measured at the terminals T_+ and T_-.

$$V_o = i_2 \cdot Z_{C_2} \quad (3)$$

$$H(s) = \frac{V_o}{V_s}, s = i\omega \quad (4)$$

Solving for $H(s)$ gives:

$$H(s) = \frac{51951s}{(s + 1.2470 \times 10^5)(s + 0.2128)} \quad (5)$$

A bode plot of the transfer function (instrument response) is shown in figure 3 below.

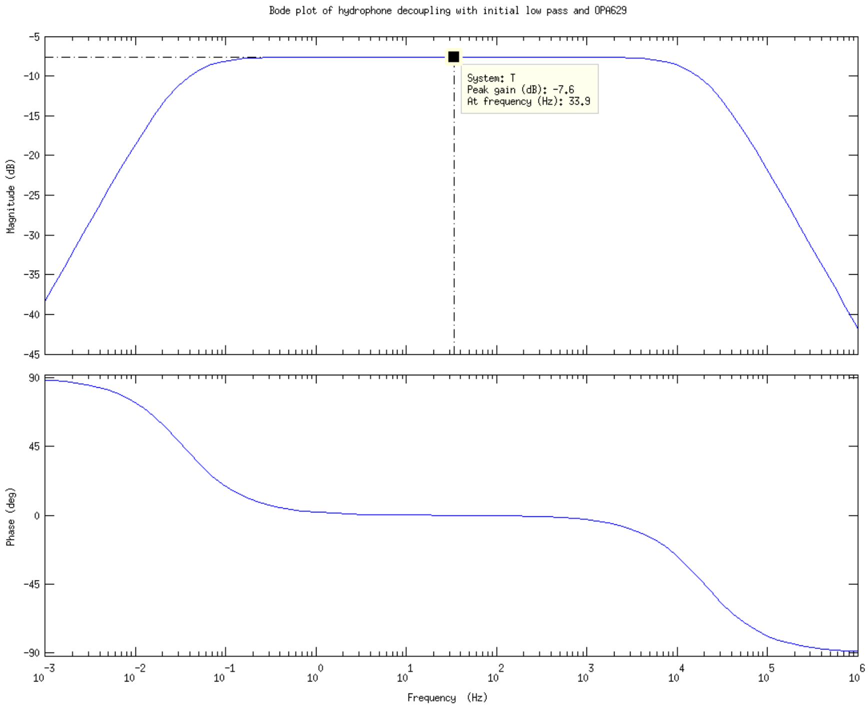

Figure 3: Bode plot of transfer function, showing peak response at 33.9 Hz.

Equation 5 can be solved by using the MATLAB script response.m (listing 1), the Bode plot in figure 3 was created with the same script.

2.1 Analysis

The initial frequency band of interest reaches from periods of 20 s (0.05 Hz) to approximately 100 Hz. The analog input part as well as the configuration of the analog-digital converter has been configured to as best and simple as possible to record the signal within this frequency band.

2.1.1 Decoupling and analog high pass filter

The transfer function is a bandpass filter with the suggested decoupling creating a high-pass filter with corner frequency at $f_{HP} = 0.0339$ Hz or a period of 29.4985 s.

2.1.2 Low pass filter and operational amplifier

To avoid aliasing due to the speed of the opamps (operational amplifiers) in the signal path; the OPA627 (see [4] and figure 1) acting as buffer with unity gain and then the OPA1632 [9] at the input terminals of the ADS1282EVM [7]. An additional low pass stage is added with a corner frequency of $f_{LP_1} = 17.189$ kHz. The speed of the opamp is in the order of Mhz so this corner frequency should be well within their capabilities.

OPA1632

The OPA1632 [9] is set up with a 1 nF capacitor in its feedback loop on the ADS1282EVM [7], this low passes the signal with an cut off frequency at $f_{LP_2} = 151.99$ kHz. Its gain bandwidth is specified to be 180 MHz, given that the gain is 1 and the load is large it should have a well behaved and near ideal response. The OPA1632-setups simplified transfer function is [8]:

$$H_{OPA1632}(s) = \frac{9.551 \times 10^5}{(s + 9.551 \times 10^5)} \tag{6}$$

OPA627

The OPA627 is used as a buffer with gain 1 to provide a high impedance input and low impedance output. It has a gain bandwidth product of 16 MHz at this gain [4]. With the first low pass filter, f_{LP_1}, well below this it should be able to relay the signal without problems. The transfer function would simply be 1 for this step. It is important that the OPA627 has well mounted bypass capacitors to avoid oscillations when configured for this low gain. An buffer designed for unity gain, like the OPA633 [5] or BUF634 [3] could prove a better choice.

OPA633

The OPA633 [5] is a strong candidate for an drop-in replacement of the OPA627, it is specifically designed to be a buffer working at unity gain. The feedback loop is internally connected. But otherwise the same wiring scheme applies. Closely mounted and well chosen bypass capacitors are still needed on the power supplies to ensure stability. The input impedance of the ADS1282EVM should be sufficient for the gain to approach unity. The OPA633 has a gain bandwidth of 260 MHz, however it requires ±15 mA current, which is about twice that of the OPA627.

2.1.3 Stability

The system, $H(s)$ from equation (5), is passive and has all its poles in the left half plane and is stable.

2.1.4 Analog-Digital Converter (ADC)

The ADS1282EVM equips its ADS1282 with a crystal with the recommended clock frequency of $f_{clk} = 4.096$ Mhz [7]. This results in a modulator output speed of $f_s = f_{clk}/4 = 1.024$ Mhz [6]. The Nyquist frequency is then:

$$f_{Nq} = 1.024 \times 10^6/2 = 512 \text{ kHz} \tag{7}$$

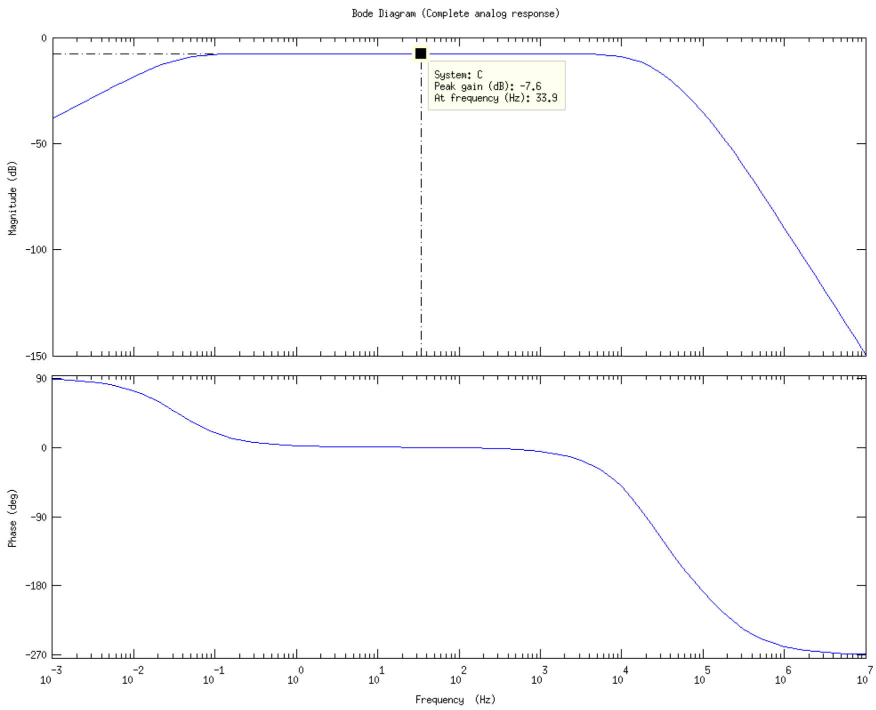

Figure 4: Frequency response of complete analog system, equation (9).

2.1.5 Anti-alias filter

The 10 nF capacitor used on the ADS1282EVM [7], results in a low-pass filter with an upper corner frequency of $f_{LP_3} = \frac{1}{2\pi \times 600 \times 10 \times 10^{-9}} = 26525.82$ Hz [1].

f_{LP_3} is thus well below f_{Nq} and any signal component above f_{LP_3} should be almost completely damped before it reaches f_{Nq}, resulting in no or very small aliasing effects.

This equation represents the upper low pass filter:

$$\begin{aligned} H_{UL}(s) &= \frac{1 \div 600 \times 10 \times 10^{-9}}{(s + 1 \div 600 \times 10 \times 10^{-9})} \\ &= \frac{1.667 \times 10^5}{(s + 1.667 \times 10^5)} \end{aligned} \tag{8}$$

The complete analog transfer function is then:

$$\begin{aligned} H_a = &\frac{8.2698 \times 10^{15}}{(s + 9.551 \times 10^5) \times (s + 1.667 \times 10^5)} \times \\ &\frac{1}{(s + 1.247 \times 10^5) \times (s + 0.2128)} \end{aligned} \tag{9}$$

The frequency response is illustrated in figure (4).

2.1.6 Output of decoupling and buffer

The output is scaled by -7.6 dB $= 0.4169 \frac{V}{V}$ which is to scale the input from $\pm 6V$ to $\pm 2.5V$, $0.4169 \approx 5/12 = 0.4167$, which is the input range of the ADS1282EVM ([7] and [6]) in bipolar mode. The slightly inaccurate scaling is due to the available (high precision) resistor values.

The impulse response of the system is plotted in figure 5:

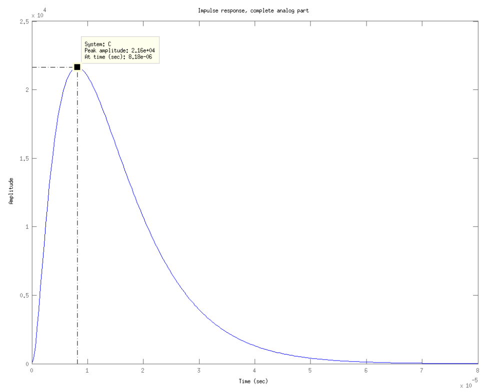

Figure 5: Impulse response of complete analog system, equation (9).

The step response, large is plotted in figure 6:

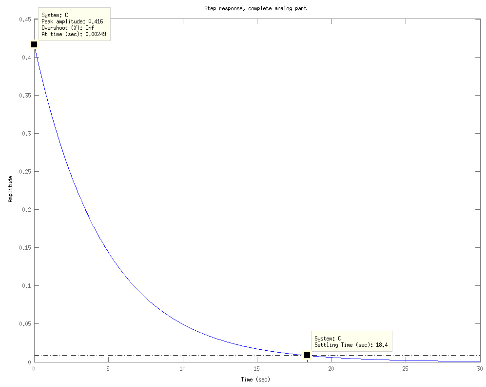

Figure 6: Step response of complete analog system, equation (9).

A zoomed in section of the high frequency response is shown in figure 7, the first tens of microseconds.

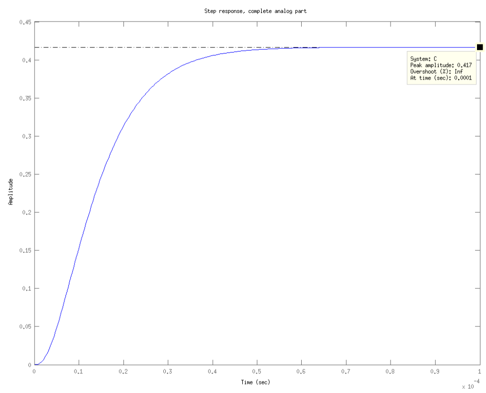

Figure 7: Step response (zoomed) of complete analog system, equation (9).

Attenuation at Nyquist frequency

The attenuation at the Nyquist frequency, 512 kHz, equation (7) using (9), is evaluated to be:

$$\begin{aligned} |H_a(512 \times 10^3 \times 2\pi)| &= 0.00023763 \\ &= -72.4821 \text{ dB} \end{aligned} \tag{10}$$

The theoretical maximum dynamic range is specified to be 130 dB[2]. But before reaching the frequencies of any signals not attenuated at (10) that wraps around to the 125 Hz passband of the digital filter (section 7) they will be additionally attenuated. The next periods will be increasingly attenuated.

Depending on the noise spectra there will be only a fraction of the noise that is in this passband. Any tones or oscillations from the electrical components

[1] Equation (3), p.15, [6]; *ADS1282 - High Resolution Analog-to-Digital Converter*

[2] Table 1, p. 13, [6]; *ADS1282 - High Resolution Analog-to-Digital Converter*

or environment are, although unlikely to be entirely in this passband, the biggest worry. Other noise is likelier to have a flatter spectra, where only a fraction of its amplitude will wrap into the 125 Hz passband.

3 Digital filter

Further the ADS1282 is configured to digitally filter the decimation output so that the final output signal of 250 Hz contains no alias below the input Nyquist sample rate (equation (7)). The built-in sinc and FIR filters are activated, see the ADS1282 datasheet [6] for details. This provides more than 140 dB of attenuation above the Nyquist frequency of the output sample rate [6].

The ADS1282 can be configured for an sample rate of as high as 4000 Hz [6]. This would be possible without changes in the analog interface, although it would come at an cost of resolution and would require adaption of the firmware of the buoy to handle the higher data rate.

High Pass Filter (HPF) The ADS1282 is configured to have an high pass corner frequency as close to 20 seconds as possible, the built in IIR HPF is set according to p. 22 in *ADS1282 - High Resolution Analog-to-Digital Converter*:

$$HPF[1:0] = 65,536 \times \left[1 - \sqrt{1 - 2\frac{\cos(\omega_n) + \sin(\omega_n) - 1}{\cos(\omega_n)}}\right] \tag{11}$$

with:

$$\omega_n = 2\pi f_{HP}/f_{DATA} \tag{12}$$

An $f_{HP} = 1/20 = 0.05$ Hz gives an $\omega_n = 0,001256637$ normalized radians. And an $HPF[1:0] = 82.3550098$ truncated to $HPF[1:0] = 82_{10} = 52_{16}$.

4 Hydrophone characteristics

The hydrophone used, Benthos AQ-18, is specified to have an flat frequency response within ±1.5dB at the band 1 - 10 kHz [2]. With a sensitivity of -171 dBv re 1 uPa at 20° C. **TODO:** Need lower corner frequency and details.. Transfer function for hydrophone:

$$H_{Hyd} = -171 \text{ dBv re 1 uPa} = 10^{8.55}\ \frac{\text{V}}{\text{uPa}} \tag{13}$$

5 Response of Digitizer

In addition to the digital filters, however very sharp should ideally be included, the output is 32 bit integers, with a range of $\left[-2^{31}, 2^{31}-1\right]$, disregarding for simplicity (**FIXME**), the transfer function for the digitization part is:

$$H_D = \frac{2^{32}}{5}\frac{\text{counts}}{\text{V}} \tag{14}$$

References

[1] Benthos. *AQ-18 suggested decoupling.*

[2] Teledyne Benthos. *Benthos Hydrophones Brochure.*

[3] Burr Brown. *BUF634 - 250mA High-Speed Buffer.* Texas Instruments. 1996.

[4] Burr Brown. *OPA627 - Precicion High-Speed Difet; Operational Amplifiers.* Texas Instruments. 1989.

[5] Burr Brown. *OPA633 - High Speed Buffer Amplifier.* Texas Instruments. 1993.

[6] Texas Instruments. *ADS1282 - High Resolution Analog-to-Digital Converter.* Revised 2010. 2007.

[7] Texas Instruments. *ADS1282EVM and ADS1282EVM-PDK User's Guide.* Revised 2011. Texas Instruments. 2009.

[8] Texas Instruments. *Fully Differential Amplifiers.* Rev. D. 2002.

[9] Texas Instruments. *OPA1632 - High-Performance, Fully-Differential Audio Operational Amplifier.* Revised 2010. 2003.

6 Attachments

6.1 response.m

Listing 1: response.m, MATLAB code for calculating instrument response

```
% Calculates instrument response
r1 = 58.33e3;  % ohm
r2 = 41.67e3;  % ohm

c1 = 47e-6;    % farad
c2 = 330e-12;  % farad

%% Transfer function of Hydrophone decoupling, including first high pass and low
%  pass filters and OPA629.

syms s i1 i2 vs;
zc1 = 1/(s*c1);
zr1 = r1;
zc2 = 1/(s*c2);
zr2 = r2;

[i1, i2] = solve ('vs = i1*(zc1 + zr1) + (i1 - i2)*zr2', ...
                  '0  = (i2 - i1)*zr2 + i2*zc2', 'i1', 'i2');
%i2 = solve ('vs = i2*(zc1 + zr1 + zr2)', 'i2');

vo = i2*sym('zc2');
H  = vo/vs;
H  = simplify(subs (H));

[n, d] = numden (H);

Tn = sym2poly(n);
Td = sym2poly(d);
T = tf(Tn, Td);


P = bodeoptions;
P.FreqUnits = 'Hz';
figure(1); clf('reset');
bodeplot (T, P);
title ('Bode plot of hydrophone decoupling with initial low pass and OPA629');

disp ('Hydrophone decoupling up to OPA629:')
zpk(T)
if isstable(T)
  disp ('System is stable.');
else
  disp ('System is unstable.');
end

%% opa1632 low pass
disp ('OPA1632. low passing:');
UOPA = zpk([], [-(1/(1047*10^-9))], 1/(1047*10^-9))
figure(4); clf('reset');
bodeplot (UOPA, P);
title ('Bode Diagram (OPA1632)');

%% upper low pass
disp ('Upper low pass (ADS1282, analog):');
U = zpk([], [-(1/(600*10*10^-9))], 1/(600*10*10^-9))
figure(2); clf('reset');
bodeplot (U, P);
title ('Bode Diagram (Upper low pass filter)');

%% final response
disp ('Final response:');
C = UOPA * U * T
figure(3); clf('reset');
```

```
bodeplot (C, P);
title ('Bode Diagram (Complete analog response)');

% attenuation at nyquist
disp ('Attenuation at Nyquist:');
a = abs(freqresp(C, 512e3*2*pi))
adb = 20*log10(a)
```