Gautebøye, rev1, Instrument response

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1 Hydrophone decoupling

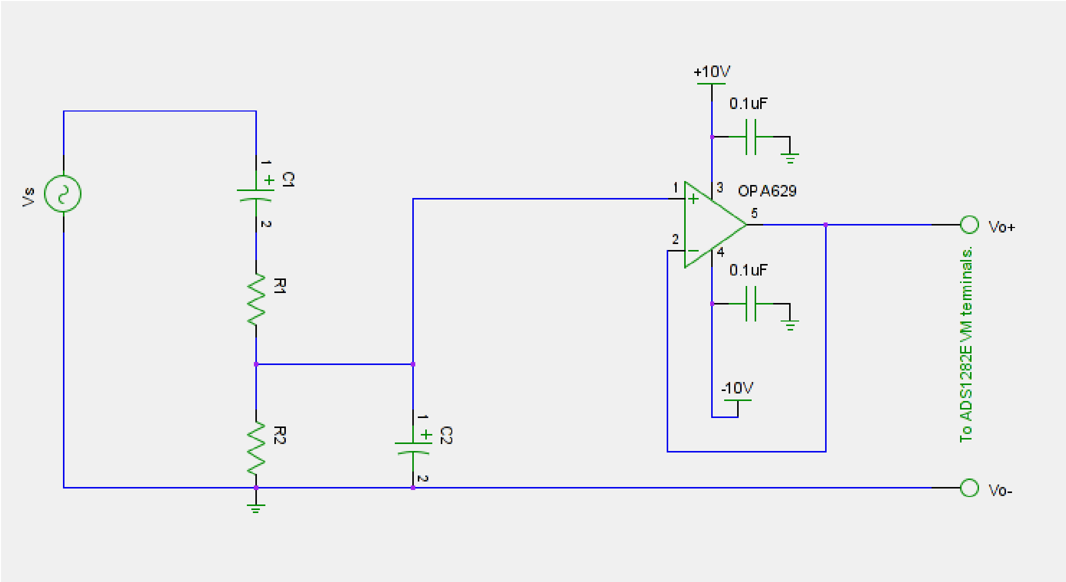

Figure 1: Hydrophone decoupling circuit

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

$$C_1 = 47\mu F$$
$$C_2 = 380pF$$
$$R_1 = 58.33k\Omega$$
$$R_2 = 41.67k\Omega$$

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}.

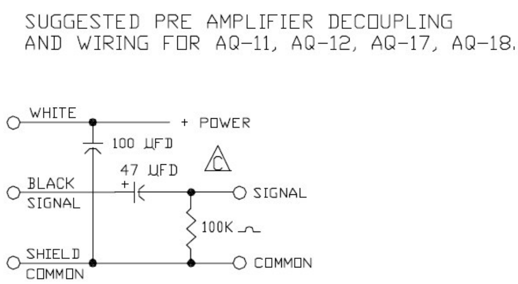

Figure 2: AQ-18 suggested decoupling and wiring [1].

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:

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

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_-.

$$V_o = i_2 \cdot Z_{C_2} \tag{3}$$

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

Solving for $H(s)$ gives:

$$H(s) = \frac{45115.36s}{(s + 1.083 \times 10^5)(s + 0.2128)} \tag{5}$$

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

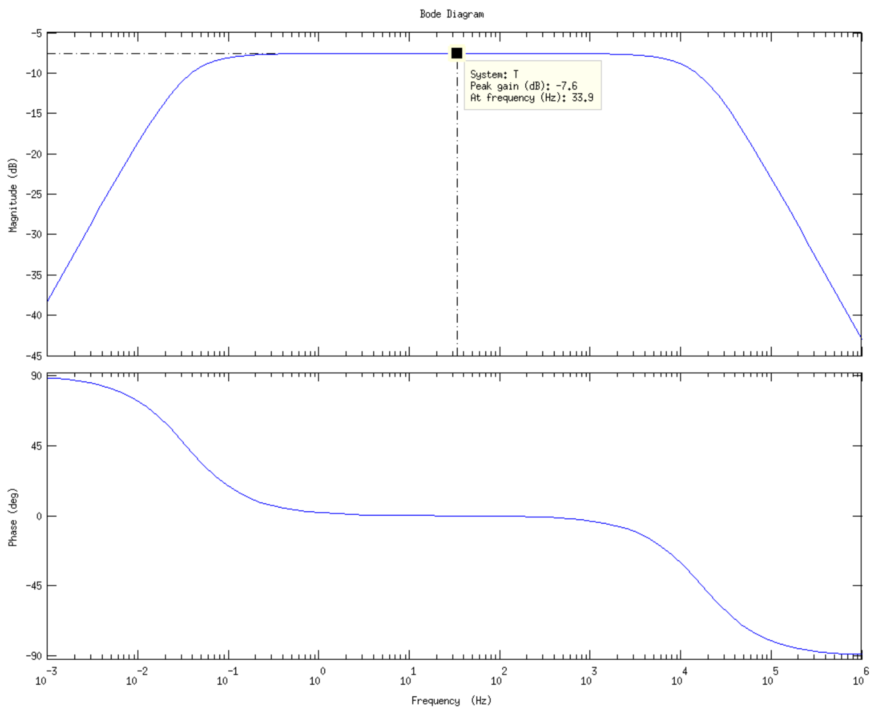

Figure 3: Bode plot of transfer function, showing peak response at 33.9 Hz.

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 [6] and figure 1) acting as buffer with unity gain and then the OPA1632 [5] at the input terminals of the ADS1282EVM [3]. 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 [5] is set up with a 1 nF capacitor in its feedback loop on the ADS1282EVM [3], 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 defined response. The OPA1632-setups simplified transfer function is [4]:

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

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 [6]. 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.

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 [3]. This results in a modulator output speed of $f_s = f_{clk}/4 = 1.024$ Mhz [2]. The Nyquist frequency is then: $f_{Nq} = 1.024 \times 10^6/2 = 512$ kHz.

2.1.5 Anti-alias filter

The 10 nF capacitor used on the ADS1282EVM [3], 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:

$$H_{UL}(s) = \frac{1 \div 600 \times 10 \times 10^{-9}}{(s + 1 \div 600 \times 10 \times 10^{-9})} = \frac{1.667 \times 10^5}{(s + 1.667 \times 10^5)} \tag{7}$$

The complete analog transfer function is then:

$$H_{analog} = \frac{7.1817 \times 10^15}{(s + 9.551 \times 10^5)(s + 1.667 \times 10^5)(s + 1.083 \times 10^5)(s + 0.2128)} \tag{8}$$

With the following frequency response:

2.1.6 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. The built-in sinc and FIR filters are activated, see the ADS1282 datasheet [2] for details. Further, the high pass filter has been set to filter at $f_{HP} = 0.05 \text{ Hz} = 20s$.

The ADS1282 can be configured for an sample rate of as high as 4000 Hz [2]. This would be possible without changes in the analog interface, although it would come at an cost of resolution.

2.1.7 Output

The output is scaled by $-7.6 \text{ dB} = 0.4169\frac{V}{V}$ 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 ([3] and [2]) in bipolar mode. The slightly inaccurate scaling is due to available resistor values.

[1] Equation (3), p.15, [2]; *ADS1282 (Datasheet)*

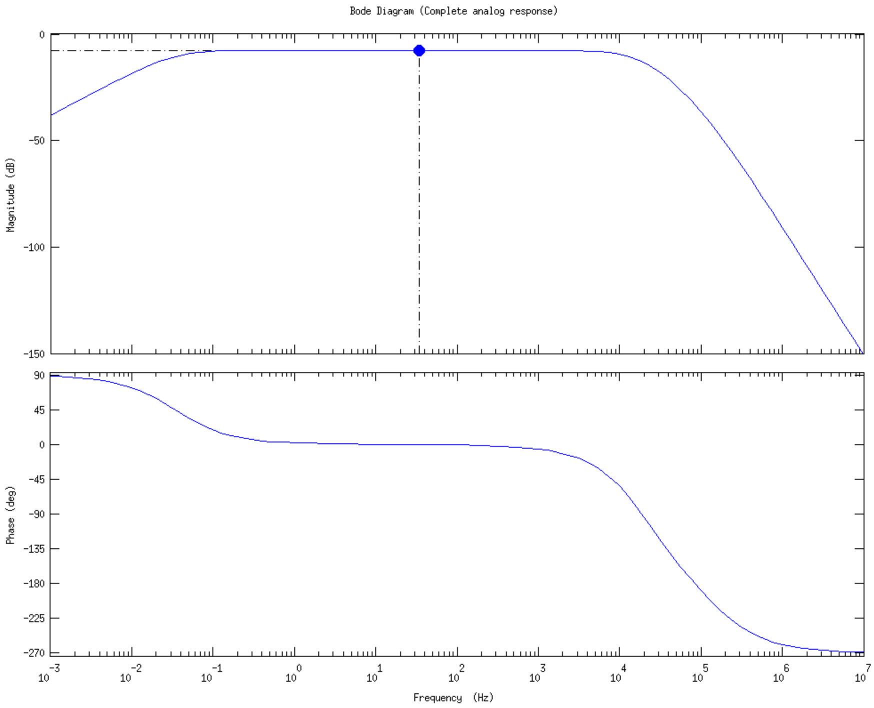

Figure 4: Frequency response of complete analog system, equation (8).

References

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

[2] Texas Instrument. *ADS1282 (Datasheet)*. rev b. Texas Instruments.

[3] Texas Instrument. *ADS1282EVM (datasheet)*. Texas Instruments.

[4] Texas Instruments. “Fully Differential Amplifiers, Rev. D”. In: (2002).

[5] Texas Instruments. *OPA1632 datasheet*.

[6] Texas Instruments. *OPA627 datasheet*. Texas Instruments.

3 Attachments

3.1 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 = 380e-12;  % farad

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;
pretty (simplify(H))
H  = simplify(subs (H));

pretty(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);

if isstable(T)
  disp ('Stable.');
else
  disp ('Unstable.');
end

zpk(T)

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

% upper low pass
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
C = UOPA * U * T;
figure(3); clf('reset');
bodeplot (C, P);
title ('Bode Diagram (Complete analog response)');
```