TAMA Modecleaner alignment error signals Version 1.4

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1 Introduction

The effects of mirror misorientations^{[1](#page-0-0)} in interferometers with three or more mirrors cannot easily be determined analytically in the general case. Therefore a Mathematica program for numerical 3-D ray tracing was written. It uses geometrical optics to find the axis of the eigenmode for a given combination of reflecting and/or refracting plane and spherical surfaces. A similar program is described and printed in Appendix E3 of MPQ243. The term 'beam' in this text refers to the geometrical axis of a beam, without taking into account the transverse shape or optical phase.

2 The modecleaner cavities

Figure [1](#page-1-0) shows a schematic view of the TAMA modecleaner cavity seen from above, together with the coordinate system adopted in this section.

The cavity consists of two flat mirrors $(M_a \text{ and } M_b)$ that are separated by a relatively short distance (20 cm), and a curved mirror M_c with radius of curvature $R = 15$ m, which is at a distance $L = 9.738 \text{ m}$ from the (center between the) flat mirrors. The beam enters through M_a and travels clockwise to M_b , M_c , M_a , etc.

M^e is the location of the photodetector behind the end mirror (in the following called 'end detector' for short), which is at a distance of 2.08 m behind mirror M_c .

There are four beams of interest leaving the cavity. The main output beam is ' Out_b ' which goes to the interferometer and the stabilization of which is the main purpose of the modecleaner. There are two beams coming from M_a : the directly reflected input beam and the beam 'Outa' which is a fraction of the cavity eigenmode. The cavity is well aligned to the incoming beam (which we consider fixed), if these two beams are perfectly superimposed. By taking two quadrant diodes with two different lens systems and appropriately demodulating their outputs, we can obtain four independent error signals, similar to the case of a simple two-mirror Fabry-Perot cavity. The longitudinal locking signal is also obtained from these two interfering beams (using the Pound-Drever-Hall scheme).

 1 To avoid confusion, we call a mirror or other component **misoriented** in this section, if its angular orientation differs from its reference orientation. The resulting movement of beams (e.g. cavity eigenmodes) will be called misalignment. An interferometer is called well-aligned if there are neither misorientations nor misalignments, i.e. components as well as beams are in their reference positions.

Figure 1: Schematic diagram of the TAMA modecleaner cavity seen from above.

In the three-mirror cavity, there are two additional degrees of freedom. They are linear combinations of movements of all three mirrors, as computed below. We will call them "neutral modes" because they have no influence on either the interference of the incoming beam with the cavity eigenmode nor on the outgoing beam. They can be used to control the spot position on the far mirror M_c . This is described in more detail below.

In the ray-tracing program, we first compute the well-aligned case (i.e., all mirrors are hit in their center) as reference. We call P_a , P_b and P_c the points where the axis of the eigenmode intersects the mirrors M_a , M_b and M_c , respectively. After misorienting one particular mirror by the small angle ε , we recompute the eigenmode axis, compare it with the well-aligned case and divide the difference by ε . The main results are:

- The shifts of the spots P_a , P_b and P_c .
- The angles γ_a , γ_b and γ_c between the beams 'Out_a', 'Out_b', 'Out_c' and their respective references. For the vertical misalignments which are considered separately, we call these angles δ_a , δ_b and δ_c , respectively. We also compute the angle γ_d (δ_d) between the directly reflected incoming beam and its reference for the case that M^a is misoriented.
- For 'Out_a' in the case of misorienting M_a , we also compute the angle $\gamma'_a = \gamma_a \gamma_d$, which is the angle between the beam leaving the cavity and the directly reflected beam, because this is the angle between the interfering wavefronts that is detected by the quadrant diode.
- As described above we finally compute the angle θ which describes the 'character' of the misalignment at the waist. It is given by $\theta^w = \arctan(\gamma'_a z_R/\Delta z_{waist})$.

The waist of the cavity eigenmode is located halfway between the mirrors M_a and

Mb. Its Rayleigh range is given by

$$
z_{\rm R} = \sqrt{\frac{L_{\rm RT}}{2} \left(R - \frac{L_{\rm RT}}{2} \right)} = 7.126 \,\mathrm{m},\tag{1}
$$

where $L_{\text{RT}}/2$ is one half of the round-trip distance:

$$
L_{\rm RT}/2 = \sqrt{L^2 + d^2/4} + d/2 = 9.8385 \,\mathrm{m}.\tag{2}
$$

3 Horizontal misalignments:

By 'horizontal' misalignments we mean that a mirror is rotated around the vertical axis, i.e. beam spots move horizontally (in the plane of the modecleaner cavity). Angles are counted as positive if a mirror is rotated clockwise, if seen from above (as in Figure [1\)](#page-1-0).

We introduce the common- and differential mode motion of mirrors M_a and M_b by defining angles α_+ and α_- . Furthermore we introduce the "neutral" mode α_n as follows:

$$
\begin{array}{c|c|c}\n & M_{\rm a} & M_{\rm b} & M_{\rm c} \\
\hline\n\alpha_{+} & \alpha_{\rm a} = \alpha_{+} & \alpha_{\rm b} = \alpha_{+} & \alpha_{\rm c} = 0 \\
\alpha_{-} & \alpha_{\rm a} = \alpha_{-} & \alpha_{\rm b} = -\alpha_{-} & \alpha_{\rm c} = 0 \\
\alpha_{\rm n} & \alpha_{\rm a} = \alpha_{\rm n} & \alpha_{\rm b} = \alpha_{\rm n} & \alpha_{\rm c} = 0.701\alpha_{\rm n}\n\end{array} (3)
$$

$$
\alpha_{+} = \frac{\alpha_{\rm a} + \alpha_{\rm b}}{2},\tag{4}
$$

$$
\alpha_{-} = \frac{\alpha_{\rm a} - \alpha_{\rm b}}{2},\tag{5}
$$

The results of the raytracing program are given in Table [1.](#page-2-0)

	Cause	P_{a}		$P_{\rm a}$		waist	Out_a		Out'_2		$\theta^{\rm w}$	
			Δx	Δz		Δz		γ a		$\gamma_{\rm a}^{\prime} = \gamma_{\rm a} - \gamma_{\rm d}$		
	α_{a}	-9.740 m \cdot α _a	$9.840\,\mathrm{m}\cdot\alpha_\mathrm{a}$			$9.739 \text{ m} \cdot \alpha_{\text{a}}$		$0.981\alpha_{\rm a}$		$-1.019\alpha_{\rm a}$	-36.72°	
$\alpha_{\rm b}$		$9.538 \text{ m} \cdot \alpha_{\text{h}}$		$-9.636 \text{ m} \cdot \alpha_{\text{h}}$		-9.739 m \cdot $\alpha_{\rm b}$		$-1.019\,\alpha_{\rm h}$		$-1.019\alpha_{\rm h}$	36.72°	
α_c		$0.288 \,\mathrm{m} \cdot \alpha_c$	$-0.291 \text{ m} \cdot \alpha_c$		$0.000 \,\mathrm{m} \cdot \alpha_c$		$2.906\alpha_c$			$2.906\alpha_c$	-90.00°	
	α_+	-0.202 m $\cdot \alpha_+$		$0.204 \,\mathrm{m}\cdot\alpha_{+}$		$0.000 \text{ m} \cdot \alpha_+$		$-0.039 \alpha_+$		$-2.039\,\alpha_{\perp}$	90.00°	
α —		-19.278 m \cdot α $-$	19.477 m $\cdot \alpha$ $-$			19.477 m $\cdot \alpha$ –		2.000α –	0.000α –		0.00°	
$\alpha_{\rm n}$		$0.000 \,\mathrm{m} \cdot \alpha_{\mathrm{n}}$	$0.000 \,\mathrm{m} \cdot \alpha_{\mathrm{n}}$			$0.000 \,\mathrm{m} \cdot \alpha_{\mathrm{n}}$		$2.000\,\alpha_{\rm n}$	$0.000\alpha_n$			
Cause							P_{c}			d. refl.		End
		P _b	P _b		Out _b				Out _c			
		Δx	Δz	$\gamma_{\rm b}$		Δx		$\gamma_{\rm c}$		$\gamma_{\rm d}$		Δx
$\alpha_{\rm a}$		$9.538 \text{ m} \cdot \alpha_{\text{a}}$	$9.637 \,\mathrm{m} \cdot \alpha_{\mathrm{a}}$	$1.019 \alpha_{\rm a}$		-0.291 m \cdot α _a		$-1.019 \alpha_{\rm a}$		$2 \cdot \alpha_{\rm a}$	$-2.411 \text{ m} \cdot \alpha_{\text{a}}$	
$\alpha_{\rm b}$		$-9.740 \,\mathrm{m} \cdot \alpha_{\mathrm{b}}$	$-9.841 \text{ m} \cdot \alpha_{\text{h}}$	$1.019 \alpha_{\rm b}$		$-0.291 \text{ m} \cdot \alpha_{\text{h}}$		$0.981\alpha_{\rm h}$		$0 \cdot \alpha_{\rm b}$	$1.749 \,\mathrm{m} \cdot \alpha_{\mathrm{b}}$	
α_c		$0.288 \,\mathrm{m} \cdot \alpha_c$	$0.291 \,\mathrm{m} \cdot \alpha_c$	$-2.906\,\alpha_{c}$		$28.596 \,\mathrm{m} \cdot \alpha_c$		$2.906\,\alpha_c$		$0 \cdot \alpha_c$	34.642 m $\cdot \alpha_c$	
α_+		-0.202 m $\cdot \alpha_+$	-0.204 m $\cdot \alpha_+$	$2.039\,\alpha_{+}$		$-0.581 \,\mathrm{m} \cdot \alpha_+$		$-0.039 \alpha_+$		$2 \cdot \alpha_+$	-0.662 m $\cdot \alpha_+$	
α		$19.278 \text{ m} \cdot \alpha$	19.477 m $\cdot \alpha$ $-$	0.000α	$0.000 \,\mathrm{m} \cdot \alpha$		-2.000α			$2 \cdot \alpha$	-4.160 m \cdot α	
$\alpha_{\rm n}$		$0.000 \,\mathrm{m} \cdot \alpha_{\mathrm{n}}$	$0.000 \,\mathrm{m} \cdot \alpha_{\mathrm{n}}$	$0.000 \alpha_n$		$19.478\,\mathrm{m}\cdot\alpha_{\mathrm{n}}$		$2.000 \alpha_n$		$2 \cdot \alpha_n$	$23.639\,\mathrm{m}\cdot\alpha_{\mathrm{n}}$	

Table 1: Results of the ray-tracing program for horizontal misalignments of the TAMA modecleaner.

4 Vertical misalignments:

In the modecleaners, the horizontal and vertical axes are not similar. The results of the raytracing program for vertical misalignments are given in Table [2.](#page-4-0) Note, for example, that a small vertical tilt β_a of the input mirror M_a (which is hit under approximately 45° from the incoming beam) causes a deflection of the reflected beam by only $\delta_d = 1.43\beta_a$ as compared to $\gamma_d = 2\alpha_a$ in the horizontal case. Another example is the tilt of M_c which, if horizontal, causes a pure angular misalignment at the waist. A vertical tilt of M_c , on the other hand, shifts the cavity eigenmode downwards parallely, without changing any angles. Angles are now counted as positive when the mirror normal moves downward from the reference direction.

As before we introduce the common- and differential mode motion of mirrors M_a and M_b by defining angles β_+ and β_- . Furthermore we introduce the "neutral" mode β_n as follows:

$$
\begin{array}{c|c|c|c|c}\n & \mathbf{M}_{\mathbf{a}} & \mathbf{M}_{\mathbf{b}} & \mathbf{M}_{\mathbf{c}} \\
\hline\n\beta_{+} & \beta_{\mathbf{a}} = \beta_{+} & \beta_{\mathbf{b}} = \beta_{+} & \beta_{\mathbf{c}} = 0 \\
\beta_{-} & \beta_{\mathbf{a}} = \beta_{-} & \beta_{\mathbf{b}} = -\beta_{-} & \beta_{\mathbf{c}} = 0 \\
\beta_{\mathbf{n}} & \beta_{\mathbf{a}} = \beta_{\mathbf{n}} & \beta_{\mathbf{b}} = \beta_{\mathbf{n}} & \beta_{\mathbf{c}} = -0.4986 \beta_{\mathbf{n}}\n\end{array} \tag{6}
$$

$$
\beta_{+} = \frac{\beta_{\rm a} + \beta_{\rm b}}{2},\tag{7}
$$

$$
\beta_- = \frac{\beta_a - \beta_b}{2},\tag{8}
$$

The results of the raytracing program are given in Table [2.](#page-4-0)

5 Degrees of freedom

The most important alignment task is to superimpose the axis of the cavity eigenmode with the axis of the incoming beam. This requires the control of four degrees of freedom. For this purpose, in the differential wavefront sensing method, we place two quadrant detectors with different lens systems in the beam reflected from Ma. The interference between the directly reflected incoming beam, which is phase modulated at an RF frequency, and the beam 'Outa' leaking out of the cavity contains enough information to lock the cavity longitudinally and to obtain alignment error signals for those four degrees of freedom that determine the superposition of the incoming beam and the cavity eigenmode.

In particular, we now assume all mirrors to be slightly misoriented and compute the combined signals which are obtained by demodulating the outputs of two quadrant detectors, one (called X_I) with $\Phi = 0^\circ$ and the other one (called X_O) with $\Phi = 90^\circ$ of extra phase shift. We scale parallel shifts Δy or Δz with the appropriate factor z_R and obtain for horizontal misalignments:

$$
X_I = -1.019 \alpha_a - 1.019 \alpha_b + 2.906 \alpha_c = -1.019 \alpha_+ + 2.906 \alpha_c, \tag{9}
$$

$$
X_Q = 1.366 \,\alpha_a - 1.366 \,\alpha_b = 1.366 \,\alpha_-, \tag{10}
$$

Cause	P _b	Out _b	P_c	Out _c	d. refl.	End
	Δy	$\delta_{\rm b}$	Δy	δ_c	$\delta_{\rm d}$	Δy
$\beta_{\rm a}$	$-3.810 \,\mathrm{m} \cdot \beta_{\rm a}$	$0.704\,\beta_{\rm a}$	$-10.661 \,\mathrm{m} \cdot \beta$	$0.704\,\beta_{\rm a}$	$1.42 \cdot \beta_{\rm a}$	$-12.125 \,\mathrm{m} \cdot \beta_{\rm a}$
$\beta_{\rm b}$	$-3.670 \,\mathrm{m} \cdot \beta_{\mathrm{b}}$	$-0.704\,\beta_{\rm h}$	$-10.661 \,\mathrm{m} \cdot \beta_{\mathrm{b}}$	$0.718 \beta_{\rm b}$	$0.00 \cdot \beta_{\rm b}$	$-12.155 \,\mathrm{m} \cdot \beta_{\mathrm{b}}$
$\beta_{\rm c}$	$-15.000 \,\mathrm{m} \cdot \beta_c$	$0.000\,\beta_{\rm c}$	$-15.000 \,\mathrm{m} \cdot \beta_c$	$0.000\,\beta_{\rm c}$	$0 \cdot \beta_c$	$-15.000 \,\mathrm{m} \cdot \beta_c$
β_+	$-7.480\,\mathrm{m}\cdot\beta_+$	$0.000\,\beta_{+}$	$-21.323 \,\mathrm{m} \cdot \beta_+$	$1.421\,\beta_{+}$	$-1.421 \cdot \beta_+$	$-24.280 \,\mathrm{m} \cdot \beta_+$
β_-	$-0.141 \,\mathrm{m} \cdot \beta$	$1.407\,\beta$	$0.000 \,\text{m} \cdot \beta_-$	$-0.014\,\beta$	$-1.421 \cdot \beta_-$	$0.030\,\text{m}\cdot\beta$
$\beta_{\rm n}$	$0.000 \,\mathrm{m} \cdot \beta_{\rm n}$	$0.000 \beta_{\rm n}$	$-13.843 \,\mathrm{m} \cdot \beta_{\mathrm{n}}$	$1.421\,\beta_{\rm n}$	$-1.420 \cdot \beta_n$	$-16.800 \,\mathrm{m} \cdot \beta_n$

Table 2: Results of the ray-tracing program for vertical misalignments of the TAMA modecleaner.

and for vertical misalignments:

$$
Y_I = 0.7035 \,\beta_a - 0.7035 \,\beta_b = 0.7035 \,\beta_-, \tag{11}
$$

$$
Y_Q = -0.524 \,\beta_a - 0.524 \,\beta_b - 2.105 \,\beta_c = -0.524 \,\beta_+ - 2.105 \,\beta_c. \tag{12}
$$

Using simple linear algebra we also find the "neutral" modes from these results. They are given in tables [\(3\)](#page-2-1) and [\(6\)](#page-3-0) above.

By inspecting tables [1](#page-2-0) and [2](#page-4-0) one finds that the main effect of these "neutral modes" is to shift the spot position on the far end mirror Mc. Hence their control is not absolutely essential for the basic function of the modecleaner. In the present situation, there is neither feedback to M_c nor is the spot position on M_c monitored. One problem with this approach might be that according to equations (9) to (12) , a small motion of M_c translates into error signals at the reflected light port which is $3...4$ times larger than the error signal for a motion of the front mirrors of comparable amplitued. This huge error signal is then fed back the the *front* mirrors which are forced to compensate Mc's misorientation although they are not very efficient actuators for that purpose. Furthermore, such feedback shifts the spot position on M_c which due to the curved mirror surface again causes the whole circle to start. It is possible that some of the present dynamic range problems may be relieved by sensing and feeding back all degrees of freedom.

Because the main effect of the "neutral modes" is to shift the spot position on the far end mirror M_c , the most straightforward approach is to sense this spot position (with a DC quadrant detector looking at the transmitted light). In tables [1](#page-2-0) and [2](#page-4-0) above, this shift is computed as Δx or Δy of point Pc, respectively.

However, the detector is located at a distance $e = 2.08 \text{ m}$ behind M_c , and the spot

position detected is hence a linear combination^{[2](#page-5-0)} of Δx or Δy of point Pc, respectively with the angle δ_{c} .

This shift of the beam spot on the photodetector is also calculated by the program and printed as "End Δy " in Tables [1](#page-2-0) and [2.](#page-4-0)

If we also scale it by the factor z_R for consistency (although the way of detection is different and hence an arbitrary scale factor appears anyway), and call the error signals thus obtained X_E and Y_E , we get

$$
X_E = -0.3383 \alpha_a + 0.2455 \alpha_b + 4.8613 \alpha_c \tag{13}
$$

$$
= -0.0464 \alpha_{+} - 0.2919 \alpha_{-} + 4.8613 \alpha_{c}, \tag{14}
$$

$$
Y_E = -1.701 \,\beta_a - 1.705 \,\beta_b - 2.105 \,\beta_c \tag{15}
$$

$$
= -1.703 \beta_{+} + 0.002 \beta_{-} - 2.105 \beta_{c}.
$$
\n
$$
(16)
$$

We see that for the horizontal direction, the additional error signal is dominated by the misorientation of M_c and could directly be fed back to that mirror, while for the vertical direction the signals are more mixed. However, they are still sufficiently linearly independent such that by inversion of the well-conditioned matrix error signals for all three mirrors can be found. This matrix inversion yields

$$
\alpha_{\rm a} = -1.6726 X_I + 1.4273 X_Q + X_E
$$
\n
$$
\alpha_{\rm b} = -1.6726 X_I - X_Q + X_E
$$
\n
$$
\alpha_{\rm c} = -0.0319 X_I + 0.1498 X_Q + 0.7015 X_E
$$
\n(17)

$$
\begin{array}{rcl}\n\beta_{a} & = & +1.6785 \, Y_I & & +Y_Q & & -Y_E \\
\beta_{b} & = & -1.6785 \, Y_I & & +Y_Q & & -Y_E \\
\beta_{c} & = & -0.0015 \, Y_I & -1.6186 \, Y_Q & & +0.4987 \, Y_E\n\end{array}\n\tag{18}
$$

where scaling factors of $1/0.301462$ and $1/0.42418$, respectively, have been applied to make the most common coefficients unity. Note that the present alignment system uses only a 2×2 matrix without X_E , Y_E , α_c and β_c .

6 Actuator transfer functions

The transfer functions from all actuators to all sensors have been measured by K. Arai. The results were fitted with LISO and are shown on the following pages.

²Many thanks to K. Arai for noting this point.

6.1 Input Mirror Yaw coil ("m1xcoil")

 α

pole0:f = 1.372388 pole0:q = 0.755353 pole1:f = 3.777486

- pole1:q = 3.455169 zero0:f = 2.508392
- zero0:q = 0.840509

6.2 Output Mirror Yaw coil ("m2xcoil")

20

pole0:q = 0.804331 pole1:f = 3.661688 pole1:q = 3.842627 zero0:f = 2.316205 $zero0:q = 0.669030$

6.3 Input Mirror Pitch PZT ("m1ypzt")

pole1:f = 20.08673

pole1:q = 16.34634

6.4 Output Mirror Pitch PZT ("m2ypzt")

10

pole0:q = 11.15256 pole1:f = 20.48184 pole1:q = 28.21300

0

pole0:f = 8.757994 pole0:q = 11.90231 pole1:f = 20.31437 pole1:q = 27.83178 zero0:f = 14.49367

zero0:q = 12.04289

6.6 Output Mirror Pitch coil ("m2ycoil")

10

 $pole0:q = 11.66273$ pole1:f = 20.58340 pole1:q = 26.94568 zero0:f = 14.29287 zero0:q = 10.79691

6.7 End Mirror Yaw Coil ("m3xcoil")

pole0:f = 1.8826161 pole0:q = 1.3989960

 Ω

pole0:f = 6.436784 pole0:q = 8.052757 pole1:f = 15.49933 pole1:q = 10.58777

6.9 End Mirror Pitch coil ("m3ycoil")

0

pole0:q = 11.48701 pole1:f = 15.65282 pole1:q = 16.45207 zero0:f = 13.19353 $zero0:q = 9.075651$

7 Calibration Factors

Both actuators and sensors have calibration factors which are not yet known. In order to find them (and test the above theroretical model), the LISO-fitted factors from the transfer functions were fitted to simple model

$$
factor = A_{\text{optical}} \times G_{\text{actuator}} \times G_{\text{sensor}}.\tag{19}
$$

where all measurements were fitted simultaneously to one set of calibration constants G_i .

For that purpose a simple C program (" prod.c ") was written which uses the Nelder-Mead Simplex algorithm to minimize

$$
\chi^2(G_1, \dots, G_m) = \sum_{i=1}^{n_{\text{data}}} w_i \left| f_i - \tilde{f}_i(G_1, \dots, G_m) \right|^2 \tag{20}
$$

where f_i are the measured factors, \tilde{f}_i their approximations from Equation [\(19\)](#page-15-0), $w_i = 1/\sigma_i^2$ are the weights of each data point derived from their standard deviation (taken from the LISO fit), and G_i the unknown calibration constants.

In fact the data set consists of two independent sets of data (for x and y), but since this does not adversely affect the fit they were nevertheless fitted all in one set.

To remove an inherent ambiguity in the model [\(19\)](#page-15-0), the calibration factor (sensitivity) for the end detector was arbitrarily fixed at unity:

$$
xe = 1,\t(21)
$$

 $ye = 1.$ (22)

Three measurements where the theoretically expected factor was zero (and the measured one was indeed small) were excluded from the fit. Hence there are 13 parameters and 24 data points.

After convergence of the Simplex algorithm, the solution was refined by a simplified Levenberg-Marquardt algorithm, and the standard errors of the parameters were computed (see Appendix of LISO manual for relevant formulae).

The results are:

8 Port usage

We use mode 7 - Advanced single chip mode. In this mode, only internal RAM (4k) and Flash-ROM (128k) are used, and all I/O lines are available. In the other modes, many of them are used as address / data lines for external memory.

8.1 Port 1

8-bit I/O, can drive LED (sink 10 mA), TTL $+$ 90 pF, Darlington.

Registers P1DDR - data direction register: 0=Input, 1=Output, default=0; P1DR - data register, default=0.

Usage Output port for 8 status LED's, via HC04 Inverter. Logic is active Low, i.e. $0 =$ LED On, 1=LED Off

P1.DR.BIT.B0 LED0 Ch1 Red

P1.DR.BIT.B1 LED1 Ch2 Red

P1.DR.BIT.B2 LED2 Ch3 Red

P1.DR.BIT.B3 LED3 Ch4 Red

P1.DR.BIT.B4 LED4 Ch5 Red

P1.DR.BIT.B5 LED5 Ch6 Red

P1.DR.BIT.B6 LED6 Green 1

P1.DR.BIT.B7 LED7 Green 2

8.2 Port 2

8-bit I/O, can drive TTL + 90 pF, Darlington. Input: programmable pull-up $(50...300\mu A =$ $16 \dots 100 \text{k}\Omega$).

Registers P2DDR - data direction register: 0=Input, 1=Output, default=0; P2DR - data register, default=0, P2PCR - pull up register, $0=$ Off, $1=$ pull-up, default=0.

Usage Output port for 8 status LED's, via HC04 Inverter. Logic is active Low, i.e. $0 =$ LED On, 1=LED Off

P2.DR.BIT.B0 LED8 Ch1 Yellow

P2.DR.BIT.B1 LED9 Ch2 Yellow

P2.DR.BIT.B2 LED10 Ch3 Yellow

P2.DR.BIT.B3 LED11 Ch4 Yellow

P2.DR.BIT.B4 LED12 Ch5 Yellow

P2.DR.BIT.B5 LED13 Ch6 Yellow

P2.DR.BIT.B6 LED14 unused

P2.DR.BIT.B7 LED15 unused

8.3 Port 3

8-bit I/O, can drive TTL $+90 \,\mathrm{pF}$, Darlington.

Registers P3DDR - data direction register: 0=Input, 1=Output, default=0; P3DR - data register, default=0.

Usage: Output port for LCD display

P3.DR.BIT.B0 LCD Data bit 4

P3.DR.BIT.B1 LCD Data bit 5

P3.DR.BIT.B2 LCD Data bit 6

P3.DR.BIT.B3 LCD Data bit 7

P3.DR.BIT.B4 LCD R/S signal

P3.DR.BIT.B5 LCD enable signal

P3.DR.BIT.B6 unused

P3.DR.BIT.B7 unused

8.4 Port 4

8-bit I/O, can drive TTL + 90 pF, Darlington. Input: programmable pull-up $(50...300\mu A =$ $16 \ldots 100 \text{k}\Omega$).

Registers P4DDR - data direction register: 0=Input, 1=Output, default=0; P4DR - data register, default=0, P4PCR - pull up register, $0=$ Off, 1=pull-up, default=0.

Usage Digital Input, pull up enabled

P4.DR.BIT.B0 via 74HC04 Inverter and Pullup - Digital Input 0

P4.DR.BIT.B1 via 74HC04 Inverter and Pullup - Digital Input 1

P4.DR.BIT.B2 via 74HC04 Inverter and Pullup - Digital Input 2

P4.DR.BIT.B3 via 74HC04 Inverter and Pullup - Digital Input 3

P4.DR.BIT.B4 via 74HC04 Inverter and Pullup - Digital Input 4

P4.DR.BIT.B5 via 74HC04 Inverter and Pullup - Digital Input 5

P4.DR.BIT.B6 - Digital Input 6

P4.DR.BIT.B7 - Digital Input 7

P4.DR.BIT.B0. . . P4.DR.BIT.B3 BCD switch 0...9

P4.DR.BIT.B4 Reset key 1=pressed

P4.DR.BIT.B5 left flipswitch 0=On

P4.DR.BIT.B6 centre flipswitch 0=On

P4.DR.BIT.B7 right flipswitch 0=On

8.5 Port 5

4-bit I/O, can drive LED (sink 10 mA), $TTL + 90 pF$, Darlington, Input: programmable pull-up $(50...300μA = 16...100kΩ)$.

Registers P5DDR - data direction register: 0=Input, 1=Output, default=0; P5DR - data register, default=0, P5PCR - pull up register, 0=Off, 1=pull-up, default=0.

Usage DAC8420 Control (2 Chips 2 bits each)

P5.DR.BIT.B0 DAC2 LD

P5.DR.BIT.B1 DAC2 CS

P5.DR.BIT.B2 DAC1 LD

P5.DR.BIT.B3 DAC1 CS

8.6 Port 6

7-bit I/O, can drive TTL $+30$ pF, Darlington,

Registers P6DDR - data direction register: 0=Input, 1=Output, default=0; P6DR - data register, default=0.

Usage unused P6.DR.BIT.B0 P6.DR.BIT.B1 P6.DR.BIT.B2 P6.DR.BIT.B3 P6.DR.BIT.B4 P6.DR.BIT.B5 P6.DR.BIT.B6

8.7 Port 7

8-bit Input / Analog Input / Analog Output. 8 channel 10-bit A/D on pins $0...7$; 2 channel 8-bit D/A on pins 6 and 7.

Registers P7DR - data register, default=0.

Usage pin $0 - A/D$ group 0 : AD channel 0 pin $1 - A/D$ group $0 : AD$ channel 1 pin $2 - A/D$ group $0 : AD$ channel 2 pin $3 - A/D$ group $0 : AD$ channel 3 pin $4 - A/D$ group 1 : AD channel 4 pin $5 - A/D$ group 1 : AD channel 5 pin $6 - A/D$ group 1 : AD channel 6 pin $7 - A/D$ group 1 : AD channel 7

8.8 Port 8

5-bit I/O , also IRQ_0, \ldots, IRQ_3 . pin 0 is I/O or $IRQ₀$, Schmitt-trigger input pin 1 is I/O or $IRQ₁$, Schmitt-trigger input pin 2 is I/O or IRQ2, Schmitt-trigger input pin 3 is I/O or $IRQ₃$, pin 4 is I/O . Output can drive $TTL + 90 pF$, Darlington.

Registers P8DDR - data direction register: 0=Input, 1=Output, default=0; P8DR - data register, default=0. IER - enables interrupt regardless of P8DDR.

Usage pin $0 - IRQ$ 0 – user pressbutton

8.9 Port 9

6-bit I/O, can drive $TTL + 30 pF$, Darlington, also used as serial port and $IRQ₄$, $IRQ₅$. pin 0 is I/O or TxD_0 , pin 1 is I/O or TxD_1 , pin 2 is I/O or RxD_0 , pin 3 is I/O or RxD_1 , pin 4 is I/O or $SCK₀$ or $IRQ₄$, pin 5 is I/O or $SCK₁$ or $IRQ₅$.

Registers P9DDR - data direction register: 0=Input, 1=Output, default=0; P9DR - data register, default=0. serial mode / IRQ overrides P9DDR.

Usage pin 1 and pin 3 (SCI1) for serial communication with PC via RS232 buffer pins 0,4 (SCI0) for synchronous serial data transfer to DAC8420

8.10 Port A

8-bit I/O, can drive TTL $+30$ pF, Darlington, Schmitt-trigger input also integrated timer unit I/O, DMA output, timing pattern output. pin 0 is I/O or TP_0 (out) or $TEND_0$ (out) or TCLKA (in), pin 1 is I/O or TP_1 (out) or $TEND_1$ (out) or $TCLKB$ (in), pin 2 is I/O or TP_2 (out) or $TIOCA_0$ (I/O) or $TCLKC$ (in), pin 3 is I/O or TP_3 (out) or $TIOCB_0$ (I/O) or $TCLKD$ (in), pin 4 is I/O or TP_4 (out) or $TIOCA_1$ (I/O), pin 5 is I/O or TP₅ (out) or TIOCB₁ (I/O), pin 6 is I/O or TP_6 (out) or $TIOCA_2$ (I/O), pin 7 is I/O or TP_7 (out) or $TIOCB_2$ (I/O).

Registers PADDR - data direction register: 0=Input, 1=Output, default=0; PADR - data register, default=0. if timing pattern output, PADDR bit must be set.

Usage pin 0 and pin 1 input from quadrature encoder switch pin 2 implicit use to cascade connect timers 0 and 1 ("finetime")

8.11 Port B

8-bit I/O, can drive LED (sink 10 mA), TTL + 30 pF , Darlington, pin 0. . . 3 have Schmitt-trigger input. also integrated timer unit I/O, DMA input, timing pattern output, A/D trigger. pin 0 is I/O or TP_8 (out) or $TIOCA_3$ (I/O), pin 1 is I/O or TP₉ (out) or TIOCB₃ (I/O), pin 2 is I/O or TP_{10} (out) or $TIOCA_4$ (I/O), pin 3 is I/O or TP_{11} (out) or $TIOCB_4$ (I/O), pin 4 is I/O or TP_{12} (out) or $TIOXA₄$ (out), pin 5 is I/O or TP_{13} (out) or $TIOXB_4$ (out), pin 6 is I/O or TP_{14} (out) or $DREQ_0$ (in), pin 7 is I/O or TP_{15} (out) or $DREG_1$ (in), $ADTRG$.

Registers PBDDR - data direction register: $0=$ Input, $1=$ Output, default=0; PBDR - data register, default=0.

if timing pattern output, PBDDR bit must be set.

Usage pin 0 (TIOCA3) has 1/2 sampling frequency as monitor.

9 Analog Inputs/Outputs

The A/D converter of the 3048 uses a stabilized +5 V reference. The nominal resolution is 10 bits, and the input range $0 \dots 5$ Volt. By averaging a few samples, an effective resolution of 11. . . 12 bits is achieved. All 8 analog input channels are used as follows:

All channels have analog 4-pole antialiasing filters with a 0.1 dB Tschebysheff characteristic and a corner frequency of 120 Hz. Their poles are

The following diagram shows the frequency response of the 120 Hz Tschebysheff filter (same for all input and output channels apart from a gain factor):

The filters are realized as 2-stage Sallen-Key circuits (see below). Furthermore, all A/D inputs of the 3048 are protected with a resistor-diode network. Channels $0...5$ have an additional resistor to the stabilized $10\,\mathrm{V}$ reference which is used for level-shifting.

The following diagram shows the input circuit for channels $0...3$:

The following diagram shows the input circuit for channels 4 and 5:

The following diagram shows the input circuit for channels 6 and 7:

The output signal comes from 2 DAC8420 D/A converters which operate with stabilized -10 V and $+10$ V reference voltages. Hence the output range is -10 V... $+10$ V. Since the required range is only $-1 \, \text{V} \ldots +1 \, \text{V}$, the output filters have a fixed gain of $1/10$. Apart from the gain, they have the same frequency response as the input filters. The following diagram shows the output filter circuit for all channels:
 $_{\rm 120~Hz}$ $_{\rm 4-pole~lowpass}$

The following diagram shows the output noise spectral density of the output filters. Above 200 Hz, it is dominated by the voltage noise of the final op-amp.

10 Digital Filters

We use recursive digital filters (IIR filters). Here some relevant formulae are summarized. A digital filter transforms an input sequence x_n , which is sampled with a sampling interval T (in our case $T = 1$ ms), into an output sequence y_n by the relationship

$$
y_i = a_0 x_i + a_1 x_{i-1} + \dots + a_n x_{i-n} - b_1 y_{i-1} - b_2 y_{i-2} - \dots - b_m y_{i-m}.
$$
 (23)

The transfer function of this filter is given by

$$
H(s) = \frac{a_0 + a_1 z^{-1} + \dots + a_n z^{-n}}{1 + b_1 z^{-1} + b_2 z^{-2} + \dots + b_m z^{-m}},
$$
\n(24)

where

$$
z = \exp(sT), \quad s = i\omega = 2\pi i f. \tag{25}
$$

Stability of the filter is determined only by the denominator of this equation. The filter is stable if and only if all complex roots of the polynomial equation

$$
z^{m} + b_{1}z^{m-1} + b_{2}z^{m-2} + \cdots + b_{m} = 0
$$
\n(26)

are within the unit circle, i.e. $|z| < 1$.

For this application we investigate three types of digital filters:

- **Perfect Integrators** These have the simple transfer function $H(s) = \alpha/s$, with gain proportional to $1/f$ and a constant $-90°$ phase shift for all frequencies. Their coeffcients can be directly computed (see Section [10.1](#page-25-0) below).
- Damped Integrators These are "stopped" integrators with the gain settling at some finite gain and the phase shift returning to zero for higher frequencies. Again their coeffcients can be directly computed (see Section [10.2](#page-26-0) below).
- General filters In order to compensate actuator resonances etc., more complicated filters are necessary. They can be computed with LISO (see Section [11](#page-27-0) below).

10.1 Perfect Integrators

The transfer function of a perfect integrator is given by

$$
H(s) = \frac{\alpha}{s},\tag{27}
$$

and sketched here:

Some relationships (omitting the unit Hz):

$$
G_{1\,\mathrm{Hz}} = \frac{\alpha}{2\pi}, \qquad \alpha = G_{1\,\mathrm{Hz}} \cdot 2\pi,\tag{28}
$$

$$
ugf = \frac{\alpha}{2\pi}, \qquad \alpha = ugf \cdot 2\pi.
$$
\n(29)

The filter is realized by the recursive relation

$$
y_i = a_0 x_i + y_{i-1},\tag{30}
$$

which has the transfer function

$$
H(s) = \frac{a_0}{1 - z^{-1}} = \frac{a_0}{1 - \exp(-sT)} \approx \frac{a_0}{1 - (1 - sT)} = \frac{a_0}{sT}.
$$
\n(31)

The approximation is valid for small frequencies $f \ll 1/T$. We see that

$$
a_0 = \alpha T,\tag{32}
$$

and hence

$$
a_0 = 2\pi T G_{1\,\text{Hz}} = 2\pi T \,\text{ugf}.\tag{33}
$$

Thanks to the simple relationship Eq. [\(30\)](#page-26-1) the filter can be realized efficiently in software, and slew-rate limiting is straightforward by limiting the increment a_0x , which is equivalent to temporarily limiting the gain.

10.2 Damped Integrators

The transfer function of a damped integrator is given by

$$
H(s) = G_{\infty} \left(1 + \frac{2\pi f_c}{s} \right) \tag{34}
$$

and shown here:

It is characterized by the gain for high frequenices, G_{∞} , and the corner frequency f_c , and can be realized by the recursive relation

$$
y_i = a_0 x_i + a_1 x_{i-1} + y_{i-1},\tag{35}
$$

which has the transfer function

$$
H(s) = \frac{a_0 + a_1 z^{-1}}{1 - z^{-1}} = \frac{a_0 + a_1 \exp(-sT)}{1 - \exp(-sT)}.
$$
\n(36)

The approximation for small frequencies by expanding the exponential function yields:

$$
H(s) \approx \frac{a_0 + a_1 - a_1 sT}{sT} = -a_1 + \frac{a_0 + a_1}{sT}.
$$
\n(37)

By comparing equations [\(34\)](#page-26-2) and [\(37\)](#page-27-1) we find

$$
a_1 = -G_{\infty}, \qquad a_0 = G_{\infty}(1 + 2\pi T f_c). \tag{38}
$$

Again this filter can be implemented efficiently in software, and slew-rate limiting can be realized by limiting the increment $a_0 x_i+a_1 x_{i-1}$. Of course the transfer function (which is defined only for linear systems) is corrupted in the case of slew-rate limiting.

11 Some notes on IIR filters on the H3048F

In this section I describe some attempts at realizing more complex IIR filters with the 3048F processor. Although theses attempts finally failed, the information may be useful for future designs.

11.1 Purpose

Looking at the transfer functions in Section [6](#page-5-1) it is clear that without special filters the unity gain frequency (UGF) will be limited to 1 or 2 Hz for each loop due to actuator resonances. Originally I had intended to obtain a UGF of around 20 Hz for better suppression of the natural motion of the mirrors. The transfer functions are well defined up to that frequency and show only a few "clean" resonances each, which need to be compensated with filters.

The plan was to use digital IIR filters inside the 3048F for that purpose. The main advantage of doing so is that the final output signal is under control of the microprocessor, enabling slew-rate limiting, a perfect "hold" function etc. If, on the other hand, the compensation filters are realized as analog external filters, the microprocessor has no way to achieve these desirable features.

In the following I will use the input mirror pitch PZT transfer function from Section [6.3](#page-8-0) as example. It has two resonances at 8.7 Hz and 20 Hz with a Q of 9.3 and 16.3, respectively. The sampling frequency is assumed to be

$$
f_{\rm samp} = 1000 \,\text{Hz}.\tag{39}
$$

The first idea to compensate these resonances is to introduce zeroes at these frequencies. However, without introducing new poles the required filter transfer function would rise as f^4 above 20 Hz, which introduces considerable noise and dynamic range problems. Hence the filter I discuss here introduces new poles of $Q = 1$ at the same frequencies of the old poles, i.e. effectively reducing the Q of the poles to 1 without changing their frequency. The following plot shows the filter transfer function. It was generated by LISO with

```
zero 8.688 9.34
zero 20.08 16.34
pole 8.688 1
pole 20.08 1
```


With such a compensation filter, a UGF of around 7 Hz could have been achieved by using an integrator active between DC and 10 Hz:

11.2 Design of IIR filters with LISO

In order to design these IIR filters, LISO was extended (version 1.81). The following input file (called m1yloop.fil)

zero 8.688 9.34 zero 20.08 16.34 pole 8.688 1 pole 20.08 1 freq log .5 200 300 iir 1000 start 5 5

produces initial values for the filter (in a file called m1yloop iir.fil) which are then refined by running LISO on that file, giving the following coefficients:

and a transfer function that is reasonably close to the desired one. However, it turns out that the required precision of the coefficients is of the order of $10^{-6} \dots 10^{-7}$, which is hard to achieve with integer arithmetic. It was found that the filter can be realized more easily by cascading two filters that compensate one pole each. Therefore in the following the compensation filter that handles the first resonance at 8.7 Hz will be investigated. It can be designed by running LISO with the following input file (called m1yl1.fil)

zero 8.688 9.34 pole 8.688 1 freq log 1 100 200 iir 1000 start

and then running the automatically produced file (called $m1y11$ **ir.fil)**. This second run of LISO uses the fitting algorithm to refine the analytical initial values, requiring stability of the filter as a constraint. The coefficients are then rounded to 16-bit integers, trying various denominators and rounding up or down, and finally selecting those integer coefficients that produce the best transfer function. The result is the following filter:

$$
y_i = \frac{8216 x_i - 16360 x_{i-1} + 8168 x_{i-2}}{8416} - \frac{-22017 y_{i-1} + 10723 y_{i-2}}{11326}.
$$
 (52)

The computed transfer function of that filter (by Equation (24)) is close to the desired one, and it is also stable. After arriving at this point I had believed that the required filters can be realized in software, after all.

11.3 Limitations of the 3048F

Experience from earlier projects and preliminary tests of this circuit and software revealed the following limitations of the 3048F for the present application:

Arithmetic with sufficient speed is limited to (signed) integer variables. Although the gnu C compiler provides a floating point library (with 32-bit single precision float variables and all usual functions), these are far too slow.

Furthermore, multiply and divide operations are limited to the hardware MULXS.W and DIVXS.W instructions, which provide a $16 \times 16 \rightarrow 32$ bit multiplication and a $32/16 \rightarrow 16$ bit division in 24 cyles $(1.5 \mu \text{sec})$ each.

Adding and subtracting 32-bit numbers is no problem (2 cyles $= 0.125 \mu$ sec for a register–register operation). However, multiply and divide operations for 32-bit numbers require many instructions and need at least 10μ sec for multiplication and 50μ sec for division.

In order to use the $32/16 \rightarrow 16$ bit DIVXS. W instruction, it is necessary to provide assembler code for the divide operation as follows:

```
inline int
mydiv (long a, int b)
{
  long result;
   asm ("divxs.w %1,%0": "=r" (result): "r" (b), "0" (a):"cc");
  return (int) result;
}
...
c = mydiv(a, b);
...
```
Otherwise the compiler will issue a subroutine call to the full 32-bit division routine. Hence the filter operations are limited to 16-bit filter coefficients and 16-bit input and output values. Intermediate products and accumulators can use 32 bit, if only add, subtract or $32/16 \rightarrow 16$ bit division operations are used.

The example [\(52\)](#page-30-0) above was designed taking into account these limitations.

Real timing data was obtained by programming the prototype, toggling digital output bits and measuring the time with an oscilloscope. The results are (for 6 input channels A/D :

Getting the data and subtracting the offset: 10μ sec for 6 channels.

Matrix operations on 16-bit integers to seperate the error signals: 118μ sec for 6 channels.

DAC output of 6 channels: $108 \mu \text{sec}$ for 6 channels.

IIR filter such as Equation [\(52\)](#page-30-0): 50 μ sec for one channel

The A/D converter was running continuously during these measurements, updating 16-bit accumulators via an interrupt handler. Once per sampling interval $(1000 \mu \text{sec})$ the accumulators are divided by the number of conversions and made available as input to the main program. This operation consumes around $10...20\%$ of the CPU time.

The above results show that with filters of the type [\(52\)](#page-30-0), operation with 6 channels at 1000 Hz should be possible. On the other hand, more complicated filters (using 32 bit or floating point variables) are out of the question.

The proposed sampling frequency of 1000 Hz may seem high for 20 Hz UGF. However, the phase shifts introduced by the the antialiasing filters (see Section [9\)](#page-22-0) and those inherent in the digital filter sampling process prohibit much slower sampling.

11.4 Performance of the IIR filter

The IIR filter [\(52\)](#page-30-0) was implemented and tested in the prototype. At first the performance seemed satisfactory, i.e. a transfer function measured with the spectrum analyzer corresponded to the prediction. More detailed tests, however, revealed a severe problem:

The filter does not react well to small input signals (e.g. a few $\%$ of the maximal amplitude).

To analyze this problem, LISO was further enhanced to simulate the integer operations of the filter. The following operations are done: For all interesting frequencies, first the maximum amplitude is determined that does not cause overflow in any intermediate result. For the example discussed here, that amplitude was 23675.[3](#page-32-0) Next, the transfer function is determined by letting the filter operate (with integer operations) on an input sine wave, and 'demodulating' the output by multiplication with $\sin(\omega t)$ and $\cos(\omega t)$ (after allowing for settling for a few periods of the input signal). The input amplitude is then reduced in steps until the transfer function deviates by 3 dB (taking into account the phase also) from the initial transfer function that was found at big amplitudes. The smallest amplitude thus found is considered the lower limit of permissible input signals.

For the present example, this lower limit turned out to be 1560 at 1 Hz (i.e. well below the 8.7 Hz resonance). Here is what happens to a 1 Hz signal of amplitude 1000 (i.e. 4.2% of the full range) in the time domain:

³The limit was caused by the second fraction of Equation [\(52\)](#page-30-0) for 100/, Hz input signals.

Clearly there is considerable crossover distortion. If the input signal amplitude is further decreased the response deteriorates rapidly and soon vanishes alltogether.

The following plot shows the minimal and maximal input amplitudes for the filter [\(52\)](#page-30-0):

The top curve (referred to the left y-axis) is the maximal signal that avoids overflow, the next curve (also referred to the left y -axis) is the minimal signal that yields a deviation of 3 dB in the transfer function, and the bottom curve (referred to the right y-axis) is the total harmonic distortion for a signal of maximal amplitude.

The cause of the problem lies in equation [\(52\)](#page-30-0), shown again here in floating-point form:

$$
y_i = 0.9762 x_i - 1.9439 x_{i-1} + 0.9705 x_{i-2} + 1.9439 y_{i-1} - 0.9467 y_{i-2}.
$$
 (53)

The coefficients of x_i , x_{i-1} and x_{i-2} nearly cancel each other. Hence the transmission of a slow signal from the input to the output relies on a signal appearing in the output. For small amplitudes, that build-up of the output signal does not take place, and the filter collapses. This effect is illustrated in the following table:

\boldsymbol{t}	in*	$_{\text{out}*}$	in	out	a_0	a_1	a_2	b_1	b_2
0.000	0.000	0.000	$\overline{0}$	$\overline{0}$	$\overline{0}$	$\overline{0}$	$\overline{0}$	$\overline{0}$	$\overline{0}$
0.001	0.628	0.613	$\mathbf 1$	$\overline{0}$	$\overline{0}$	$\overline{0}$	$\boldsymbol{0}$	$\overline{0}$	$\boldsymbol{0}$
0.002	1.257	1.198	$\mathbf{1}$	$\overline{0}$	$\boldsymbol{0}$	-1	$\overline{0}$	$\overline{0}$	$\overline{0}$
0.003	1.885	1.755	$\overline{2}$	$\overline{0}$	$\mathbf 1$	-1	$\overline{0}$	$\overline{0}$	$\overline{0}$
0.004	2.513	2.286	3	$\overline{0}$	$\overline{2}$	-3	$\boldsymbol{0}$	$\overline{0}$	$\overline{0}$
0.005	3.141	2.793	3	$\overline{0}$	$\overline{2}$	-5	$\mathbf{1}$	$\overline{0}$	$\overline{0}$
0.006	3.769	3.278	$\overline{4}$	$\overline{0}$	3	-5	$\overline{2}$	$\overline{0}$	$\overline{0}$
0.007	4.397	3.742	$\overline{4}$	$\overline{0}$	3	-7	$\overline{2}$	$\overline{0}$	$\overline{0}$
0.008	5.024	4.186	$\bf 5$	$\boldsymbol{0}$	$\overline{4}$	-7	3	$\overline{0}$	$\boldsymbol{0}$
0.009	5.652	4.613	6	$\overline{0}$	$\overline{5}$	-9	3	$\overline{0}$	$\overline{0}$
0.010	6.279	5.023	$\sqrt{6}$	$\overline{0}$	$\overline{5}$	-11	$\overline{4}$	$\overline{0}$	$\overline{0}$
0.011	6.906	5.419	$\overline{7}$	$\boldsymbol{0}$	66	-11	$\overline{5}$	$\overline{0}$	$\overline{0}$
0.012	7.533	5.801	8	$\overline{0}$	$\overline{7}$	-13	$\overline{5}$	$\overline{0}$	$\boldsymbol{0}$
0.013	8.159	6.171	8	$\overline{0}$	$\overline{7}$	-15	6	$\overline{0}$	$\boldsymbol{0}$
0.014	8.785	6.531	9	$\boldsymbol{0}$	8	-15	$\overline{7}$	$\overline{0}$	$\overline{0}$
0.015	9.411	6.881	9	$\overline{0}$	8	-17	$\overline{7}$	$\overline{0}$	$\boldsymbol{0}$
0.016	10.036	7.224	10	$\mathbf{1}$	9	-17	8	$\overline{0}$	$\overline{0}$
0.017	10.661	7.559	11	$\mathbf{1}$	10	-19	8	$\mathbf{1}$	$\overline{0}$
0.018	11.286	7.889	11	$\boldsymbol{0}$	10	-21	9	$\mathbf{1}$	$\boldsymbol{0}$
0.019	11.910	8.214	12	$\mathbf{1}$	11	-21	$10\,$	$\overline{0}$	$\overline{0}$
0.020	12.533	8.536	13	$\mathbf{1}$	12	-23	10	$\mathbf{1}$	$\overline{0}$
0.021	13.156	8.855	13	$\overline{0}$	12	-25	11	$\mathbf{1}$	$\overline{0}$
0.022	13.779	9.173	14	$\mathbf{1}$	13	-25	12	$\overline{0}$	$\overline{0}$
0.023	14.401	9.491	14	$\mathbf{1}$	13	-27	12	$\mathbf{1}$	$\overline{0}$
0.024	15.023	9.808	15	$\mathbf{1}$	14	-27	13	$\mathbf{1}$	$\boldsymbol{0}$
0.025	15.643	10.127	16	$\overline{0}$	15	-29	13	$\mathbf{1}$	$\overline{0}$
0.026	16.264	10.448	16	$\boldsymbol{0}$	15	-31	14	$\overline{0}$	$\overline{0}$
0.027	16.883	10.772	17	$\mathbf{1}$	16	-31	15	$\overline{0}$	$\overline{0}$
0.028	17.502	11.099	18	$\mathbf{1}$	17	-33	15	$\mathbf{1}$	$\overline{0}$
0.029	18.121	11.430	18	$\boldsymbol{0}$	17	-34	16	$\mathbf{1}$	$\boldsymbol{0}$
0.030	18.738	11.765	19	$\mathbf{1}$	18	-34	17	$\overline{0}$	$\boldsymbol{0}$

t is the time in msec, "in∗" a (floating point) input signal of 1 Hz and amplitude 100 and "out∗" the output signal of the filter [\(53\)](#page-34-0), implemented in floating point. The rest of the table refers to the filter [\(52\)](#page-30-0), implemented in integer arithmetic. "in" is the input signal, "out" the output. The columns labelled " a_0 " to " b_2 " show the individual contributions to the output signal, as given in Equation [\(52\)](#page-30-0). In the table, they don't

add up exactly, because the output signal is computed as

$$
y_i = \frac{8216 x_i - 16360 x_{i-1} + 8168 x_{i-2}}{8416} - \frac{-22017 y_{i-1} + 10723 y_{i-2}}{11326},
$$
(54)

whereas the table entries are shown as

$$
\frac{8216 x_i}{8416}, \frac{-16360 x_{i-1}}{8416}, \frac{8168 x_{i-2}}{8416}, \frac{-22017 y_{i-1}}{11326}, \frac{10723 y_{i-2}}{11326}, \tag{55}
$$

respectively. Due to the near cancellation of the coefficients $a_0 + a_1 + a_2$, the output signal fails to build up in the presence of rounding errors. The DC gain, however, relies on an output signal building up, and hence the filter completely fails.

11.5 Attempts to improve the IIR filter

After the failure described in the previous section, I have tried to improve the filter by removing the DC gain, i.e. making an IIR filter that implements the transfer function

$$
H'(s) = 1 - H(s),\tag{56}
$$

such that the real output signal is found by

$$
output = input + H'(s)(input). \tag{57}
$$

The hope was that a DC signal would be transferred directly to the output, bypassing the IIR filter, which would only become active for signals near the resonance frequency. The following plot shows the transfer function $H'(s)$:

LISO was extended once more to yield the following filter:

$$
y_i = \frac{-778 x_i + 0 x_{i-1} + 778 x_{i-2}}{32767} - \frac{-22053 y_{i-1} + 10741 y_{i-2}}{11345}.
$$
 (58)

Although the performance for DC is now good, as expected, the performance remains poor for signals near the resonance. The sharp change of the minimal input signal is caused by the somewhat arbitrary limit of 3 dB deviation in the transfer function, which is not at all reached for signals far enough from the resonance.

I have concluded that it is not possible to implement compensation filters with 16-bit integer arithmetic for the present project.

Since the design of IIR filters itself is no problem, there is no fundamental problem in using them. For future designs I would try to find a suitable processor which has floating point arithmetic (e.g. a DSP or something like a 80486DX with 64/80-bit floating point coprocessor). It is, however, questionable, if such processors have all the convenient I/O features of the 3048F microcontroller, that are used for controlling the LCD, input switches, serial connection to the DAC etc.