Appendix A

Hardware

This appendix presents some pertinent background information on the frequency synthesizers and filters we use in our experiment.

A.1 Frequency Synthesizers

We use SRS [91] DS345 30 MHz synthesized function generators as our signal generators in our research. The DS345 generates a signal using direct digital synthesis (DDS). The advantage of DDS is the ability to smoothly vary the phase of the synthesized output. There are three types of frequency synthesizers commonly used today: direct-analog synthesis, indirect synthesis, and direct-digital synthesis.

A.1.1 Signal Synthesis

Direct-analog synthesis function generators use collections of multipliers, dividers, and other mathematical manipulations to produce the desired output frequency. Sweeping the frequency of such a device results in large jumps of the phase as different mathematical manipulations must be performed to produce the frequencies f and $f + \Delta f$ from the reference.

Indirect synthesis function generators use a phase-lock-loop (PLL) in conjunction with a voltage controlled oscillator (VCO). The output of the VCO is phase compared with some reference signal. Deviations in the phase of the VCO are corrected for by changing the applied voltage. These fine steps in the voltage degrade the phase noise performance of the synthesizer.

The only oscillator component in direct-digital synthesis, sometimes referred to as a numerically-controlled oscillator (NCO), is the external reference frequency. Although there are various implementations of DDS, the conventional DDS system consists of three major components: a phase accumulator, a look-up table, and a digital-to-analog converter (DAC)

Using the reference signal as a clock, the phase accumulator computes an address for the look-up table based on the frequency of the desired signal and the previous address. The look-up table contains a waveform stored in random-access memory (RAM). The waveform can be sinusoidal, triangular, square, or even an arbitrary waveform downloaded from a computer. The DAC converts the digital values in the look-up table into analog voltage levels. A high-order low-pass filter removes any high frequency harmonics from the output of the DAC due to the discreteness of the look-up table. A collection of amplifiers and attenuators produce the desired signal level.

A frequency sweep, such as the one performed in our research, is accomplished by instructing the phase accumulator to accumulate phase at an additional rate proportional to t^2 over the nominal rate of t. This is done by utilizing another look-up table. The waveforms in this table are the modulation waveforms such as a linear ramp, logarithmic ramp, sinusoid, triangle, square, or arbitrary.

Ignoring small effects, the output of the frequency synthesizer when modulated with a linear frequency ramp will have a smooth phase. The small effects are phase spurs. They appear as small glitches, or ringing, at each of the DAC transitions. Since these glitches are high in frequency they are filtered out by the high-order low-pass filter used to transform the jagged square looking sine wave into a smooth sine wave.

A.1.2 Noise Performance

The specified noise performance of one of these instruments is less than "-55 dBc in a 30 kHz band centered on the carrier, exclusive of discrete spurious signals" [91]. This is 1.78×10^{-3} radians, or 2.8×10^{-4} cycles in a 30 kHz band. If we assume the phase noise is white in this band then the phase noise in a 1 Hz band would be 1.63 μcycles/$\sqrt{\text{Hz}}$.

The measured phase noise will be a combination of any intrinsic instrumental phase noise, amplitude noise, and frequency noise in the output. We can make a "DC" phase noise measurement to use as a comparison with our AC phase meter. The output of the frequency synthesizer is put into both the LO and RF ports of a mixer. We have a low-pass filter at the output of the mixer. The resulting nearly DC level is a measure of the noise in the system. The voltage noise we measure for our function generator is fairly independent of signal frequency. The voltage noise level for a 1 MHz signal is shown in Figure A.1. The level is very nearly 200 nV/$\sqrt{\text{Hz}}$.

A.2 Filters

We use a collection of filters in our experiment. The purpose of most of our filters is to remove unwanted mixing products from the outputs of our frequency mixers. The single most important filter

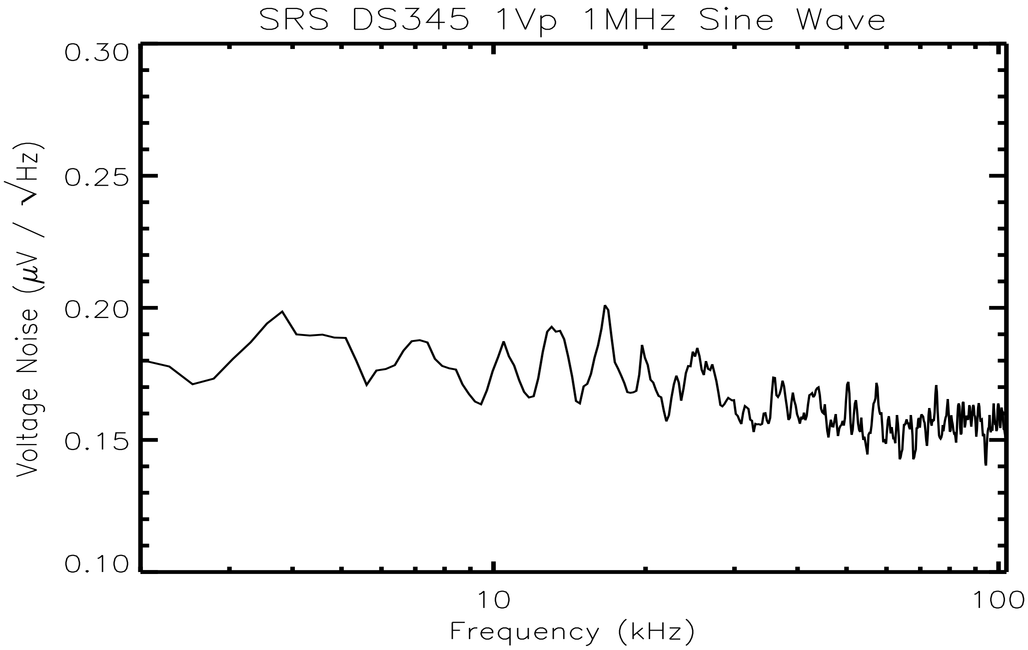

Figure A.1: This data was produced by injecting a signal from our frequency synthesizer into both the LO and RF ports of a mixer. The low-pass IF output was sent to our FFT machine. After correcting for gain and the factor of $\sqrt{2}$ contribution in noise we arrive at this plot. The voltage noise at a frequency of 10 kHz is roughly $180\,\mathrm{nV}/\sqrt{\mathrm{Hz}}$.

in our experimental setup is the crystal filter used in the frequency generation technique explained in §4.2. At the output of each demixer we have a low-pass filter. There is one demixer for the phasemeter, and another for the laser communication output. Transfer function diagrams of these two mixers (which were used interchangeably) are in Figures A.2 and A.3. The transfer function of the low-pass filter used for the acoustical stabilization loop is shown in Figure A.4. Instead of using a bandpass filter at the output of PDc we use a high-pass filter, see Figure A.5. In LISA the idea of using a bandpass filter is to remove the roughly 100 kHz ranging sidetones from the subcarrier, and vice-versa. In our case the sidetone is 10 kHz, and filtering of this signal, while retaining the subcarrier, is rather difficult with a bandpass filter at the output of PDc.

A.2.1 Narrowband Crystal Filter

A narrowband crystal filter is at the heart of our frequency generation technique. The requirements on the filter are based on the operating bandwidth of our AOM and the smallest Doppler frequency we wish to investigate. The minimum Doppler frequency we wish to investigate is in the tens of kilohertz. The center frequency of our AOMs is 80 MHz. The bandwidth of our AOMs appears to be on the order of 40 MHz.

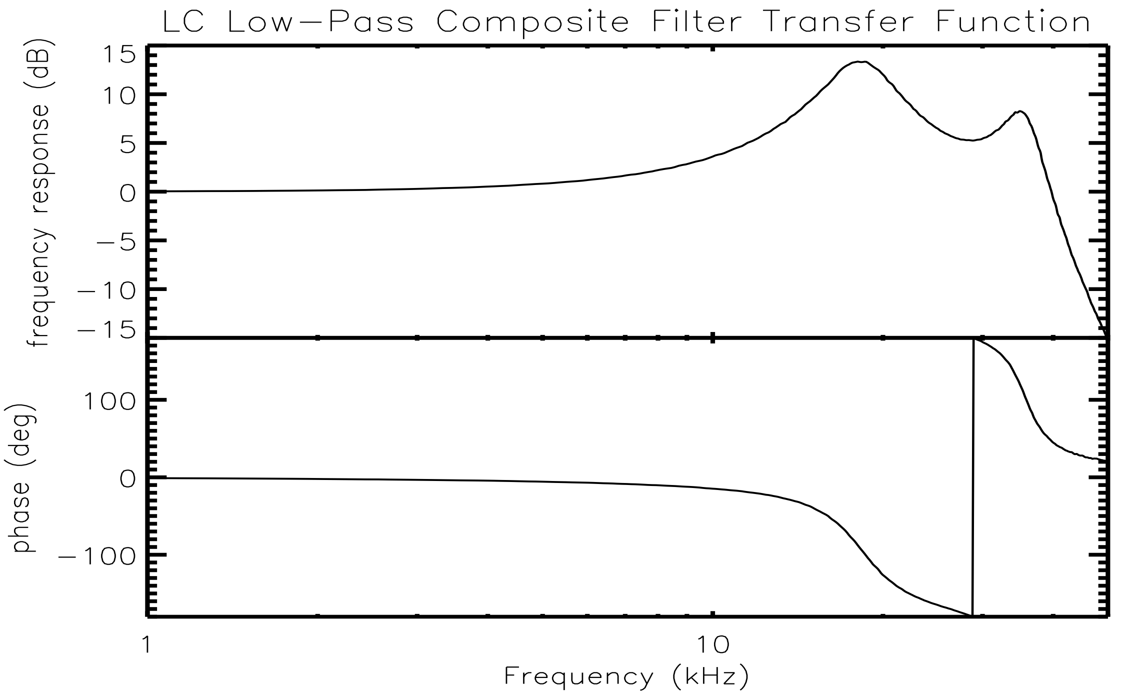

Figure A.2: A composite low-pass filter made up of a 4-pole 30 kHz low-pass, a 2-pole 1 MHz low-pass, and a 2-pole 20 MHz low-pass. The frequency response of the filter is relatively flat at a level of -60 dB for frequencies above 2 MHz. The attenuation level declines at frequencies above 100 MHz.

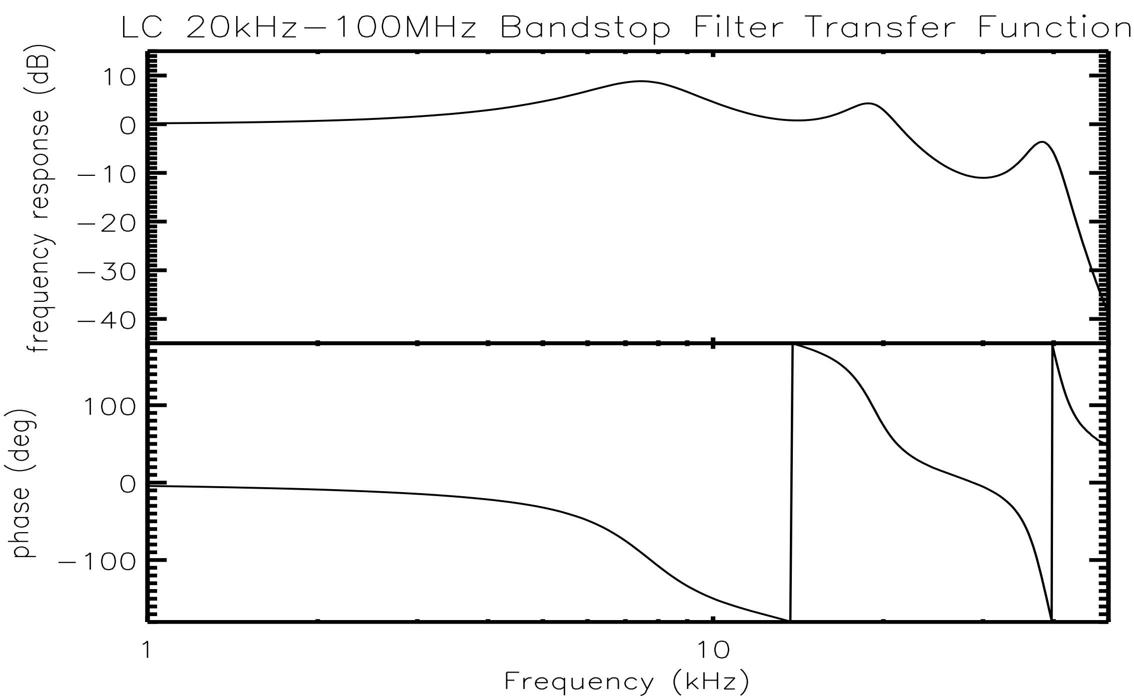

Figure A.3: Our "monster" bandstop filter contains 16-poles. The attenuation level in the flat bandstop above 75 kHz is nearly -90 dB. The attenuation level declines at frequencies above 90 MHz.

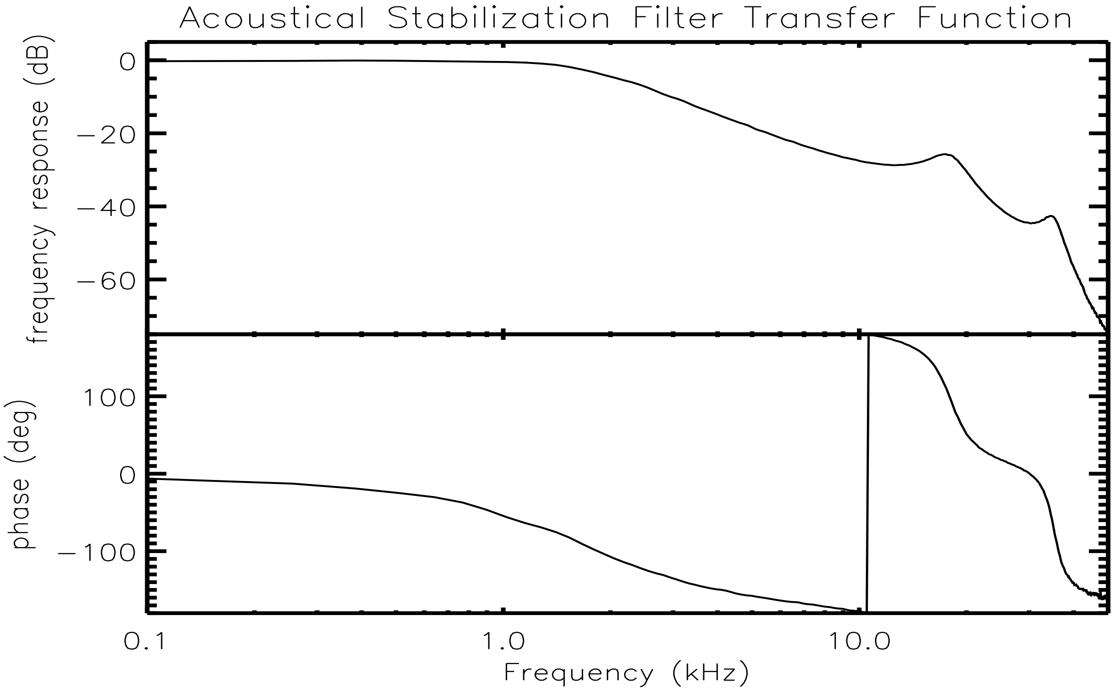

Figure A.4: The error signal in our acoustical stabilization loop is the phase difference between the input signal and the output signal of our interferometer. This error signal is created by mixing these two signals. This low-pass filter is a composite made of the filter shown in Figure A.2 and a 1.6 kHz 2-pole low-pass filter.

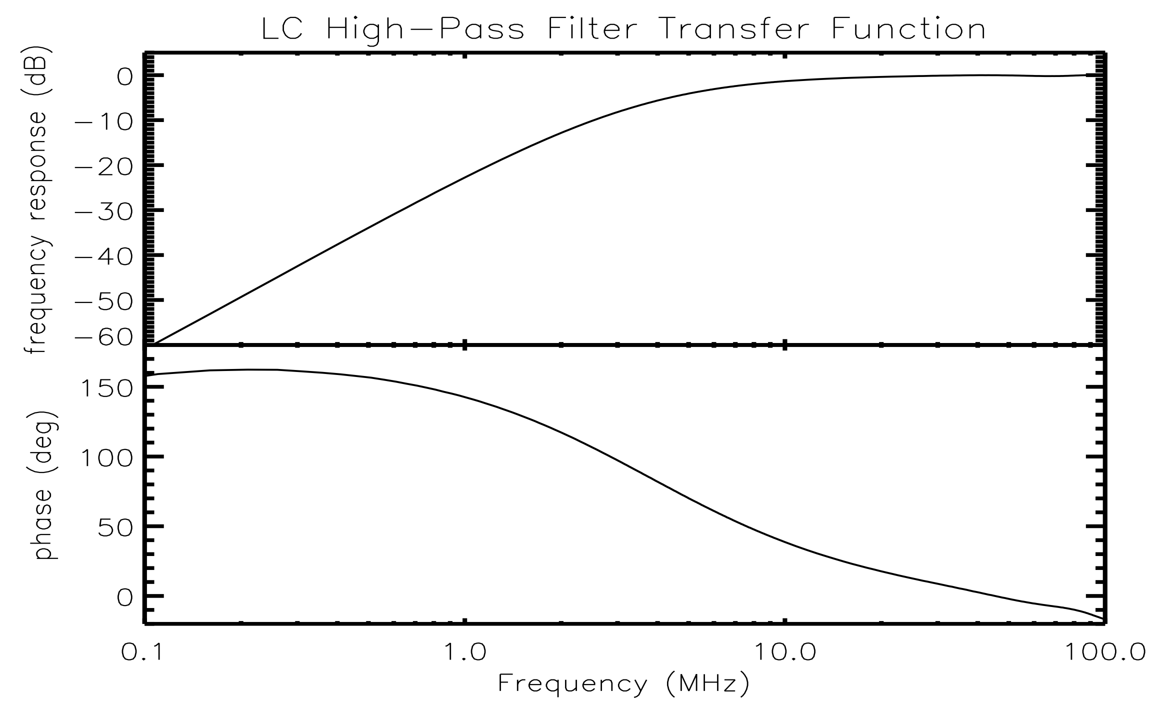

Figure A.5: This high-pass filter is used instead of a bandpass filter for the communication signal. The requirement on this filter is to remove any DC signal produced from the heterodyne process. The 30 MHz subcarrier is easily in the passband of this 10 MHz high-pass filter.

The signal from our "high frequency source," $f \approx 70\,\mathrm{MHz}$, is mixed with the output from our signal frequency synthesizer which ranges in frequency from $\delta f = 25\,\mathrm{kHz}$ to $20\,\mathrm{MHz}$. The output of the mixer contains f, δf, $f \pm \delta f$, $f \pm 2\delta f$, and higher harmonics. The purpose of the crystal filter is to remove the power at all unwanted frequencies. Therefore the width of the filter has to be on the order of minimum δf.

We never sweep δf over more than the bandwidth of the crystal filter. When we step δf from one frequency to another, i.e., when switching between examining a Doppler of 50 kHz and a Doppler of 100 kHz, we also step our high frequency source to keep the mixing product $f + \delta f$ at the center of the crystal passband.

We use a monolithic crystal filter produced by ECS Inc [15]. The specifications for this crystal are in Table A.1. Figure A.6 contains the measured transfer function for our implementation of the ECS crystal filter.

Table A.1: 70 MHz Monolithic Crystal Filter
(Operating Temperature −20 to +70° C) [15]

center frequency	number of poles	passband	ripple max.	insertion loss
70 MHz	4	±7.5 kHz	1.0 dB	4.0 dB

stopband attenuation	guaranteed attenuation	terminating impedance
±35 kHz 40 dB	±910 kHz 70 dB	2000 Ω // −1.0 pF

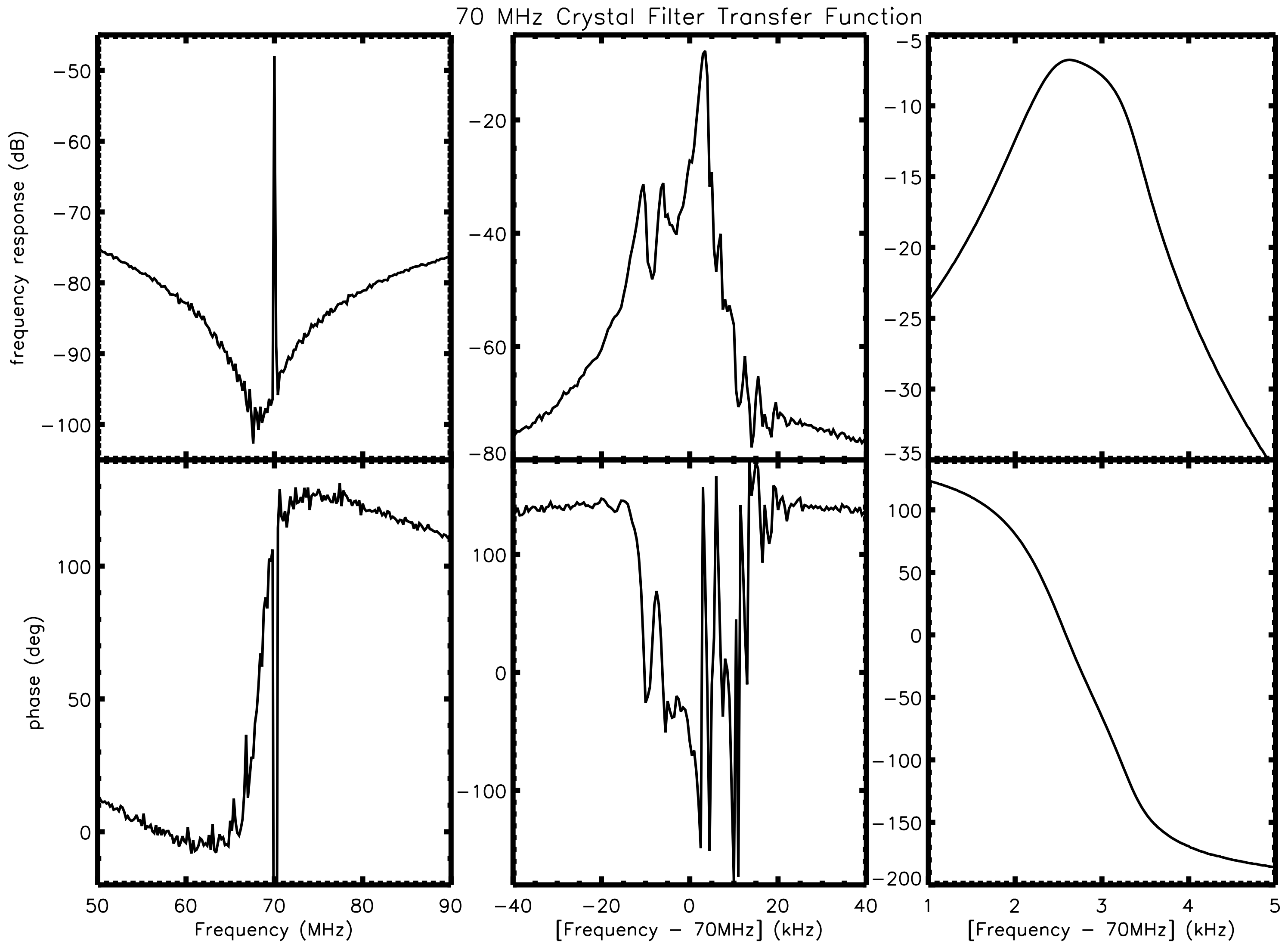

Figure A.6: Our implementation of the ECS crystal filter has passband of about 1 kHz with an attenuation level of about 7 dB. The ripple in the passband is a few dB. The phase changes smoothly over this passband from 180° to −180°. This variation in amplitude and phase throughout the passband effects the apparent sweep rate of the signal as discussed in §4.3.5.