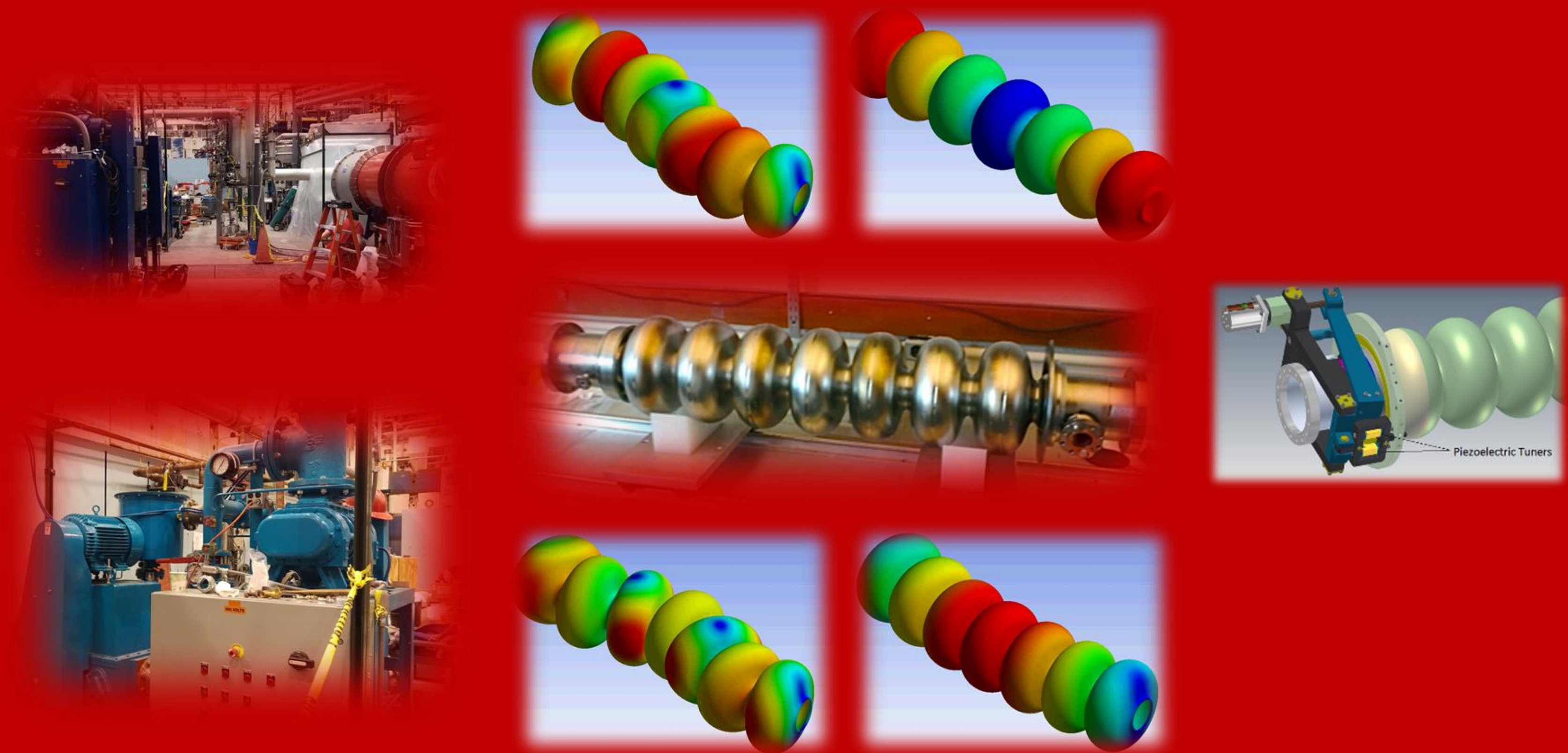

Microphonics Compensation for Low Bandwidth SRF Cavities in the CBETA ERL

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Overview

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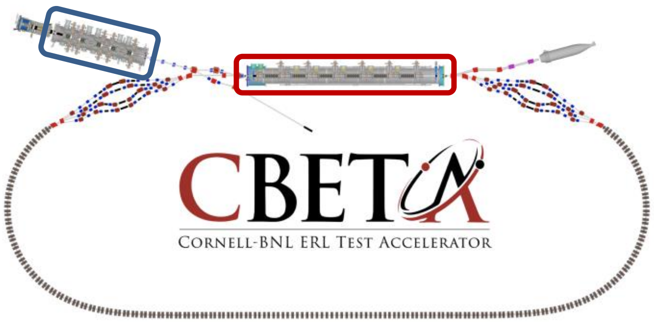
3. Tuner Response
4. Compensation Techniques
5. Results
6. Summary

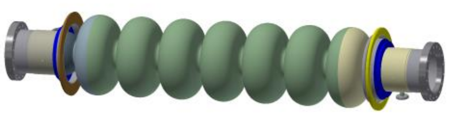
Introduction

The **Cornell-BNL ERL Test Accelerator** is a 4-turn energy recovery Linac with a FFAG return loop.

Injector Linac
5 2-cell SRF cavities
$Q_L \sim 5 \cdot 10^4$

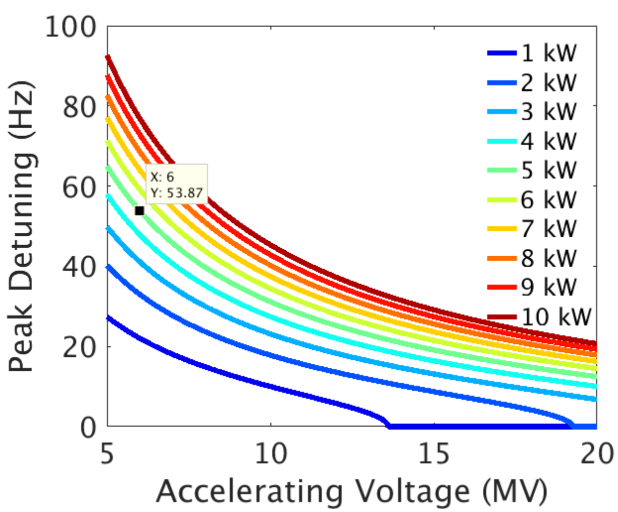


Main Linac
6 7-cell SRF cavities
$Q_L \sim 6 \cdot 10^7$



The injector cavities are powered by Klystrons capable of delivering a forward power of 100kW.

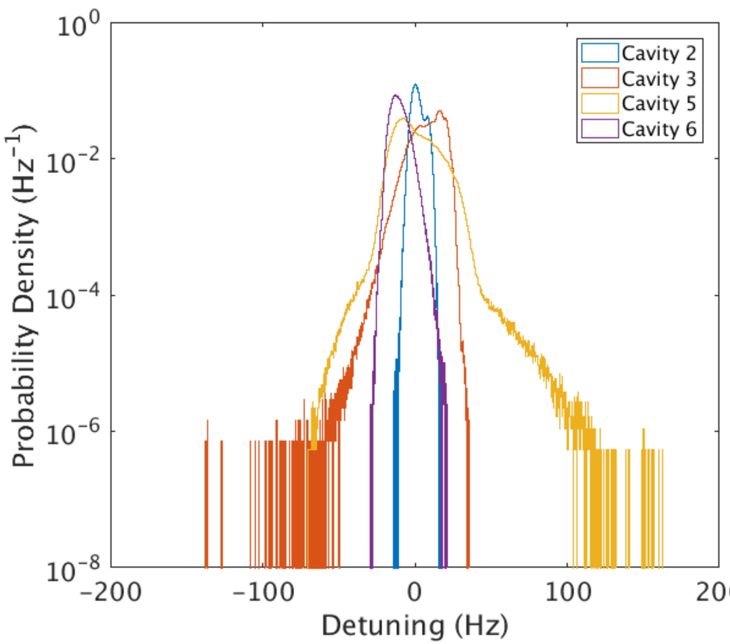
The Main Linac cavities are powered by solid state amplifiers capable of delivering a forward power of 5kW.

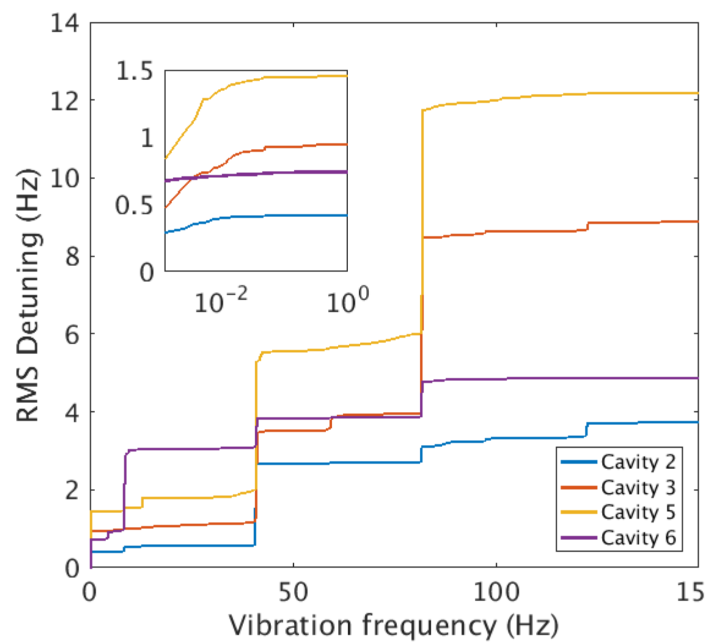


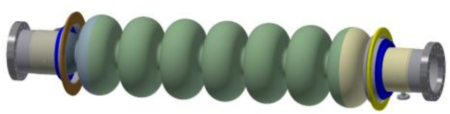
The peak detuning of the cavity must be less than 54 Hz in order to sustain a cavity voltage of 6MV using a power amplifier capable of delivering 5kW.

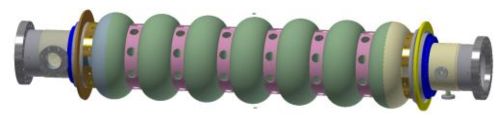
We're going to focus on the Main Linac cavities for the rest of this presentation.

Microphonics Measurements









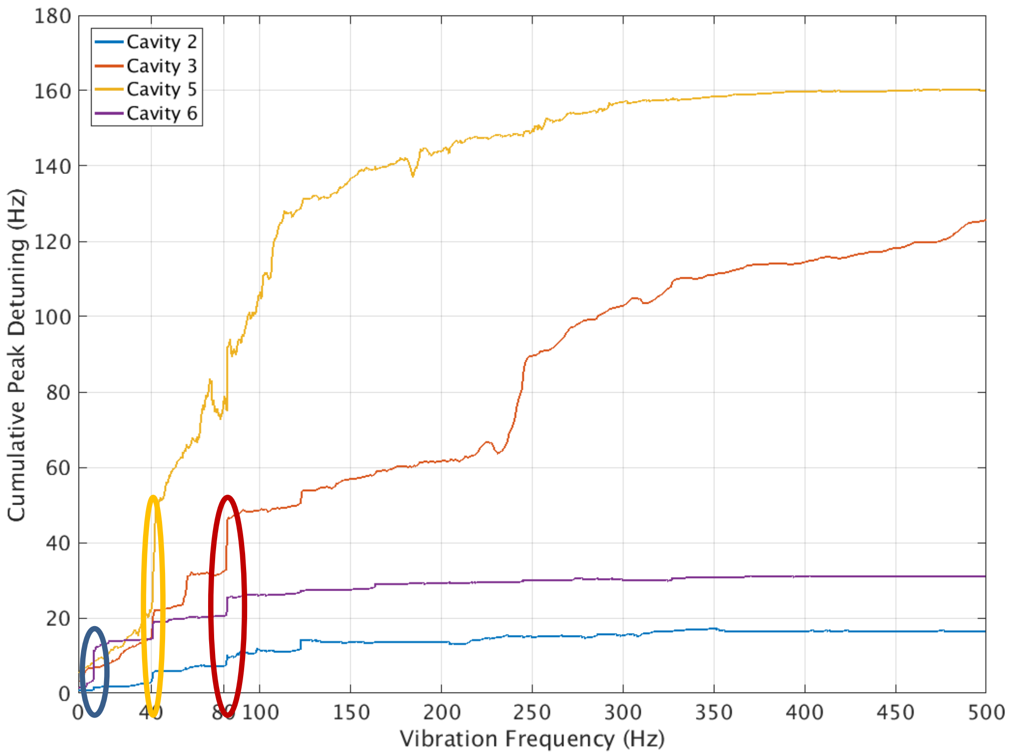
Cavity Number	3	5	2	6
Measurement Voltage (MV)	1	0.5	4	4
Peak Detuning (Hz)	137	163	18	33
Major Vibration Frequencies (Hz)	40, 80	40, 80	40	8, 40, 80

Peak Events

Microphonics measurements are commonly visualized in two ways:

- **Histograms**: These show peak detuning values without any frequency information.
- **Spectrum Plots** : These show the average contribution of different frequency components.

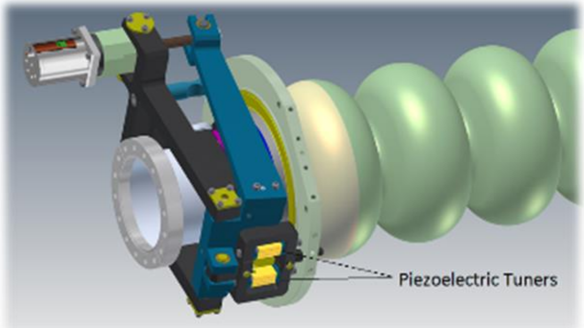
Which frequencies actually contribute to peak events?

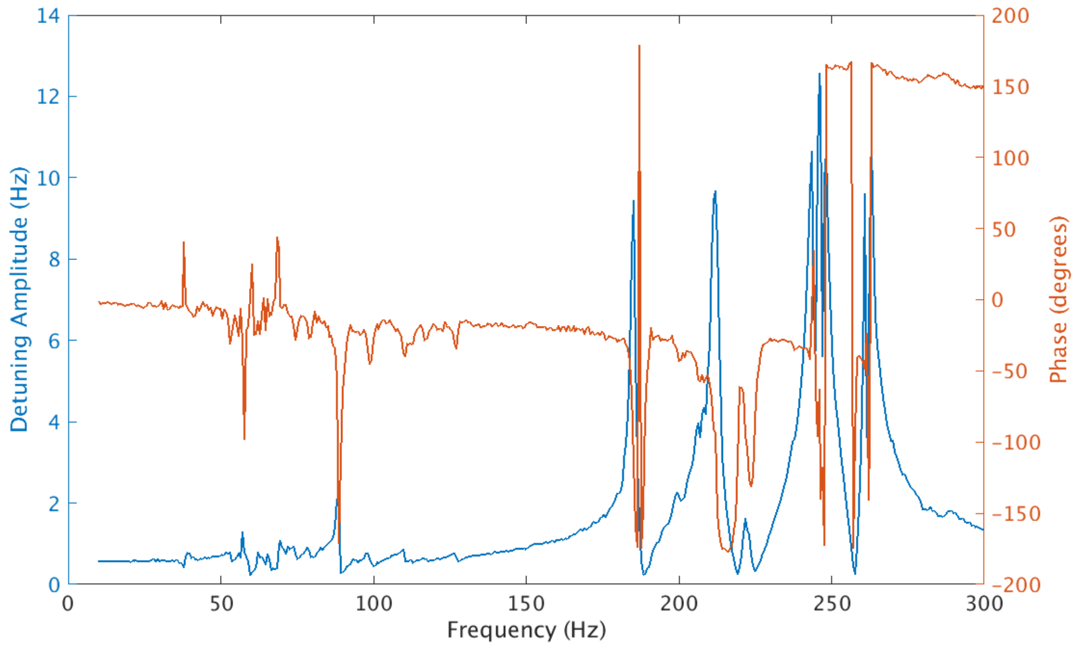


- The same frequencies appear for stiffened cavities as in the plot of RMS detuning.
- However, some peak events are very wide-band in the un-stiffened cavities and would be much harder to compensate.

Tuner Response

Fast tuners driven by piezoelectric actuators are use to compensate for detuning. Assuming that the tuner is a **Linear Time Invariant** system, we can measure a transfer function.





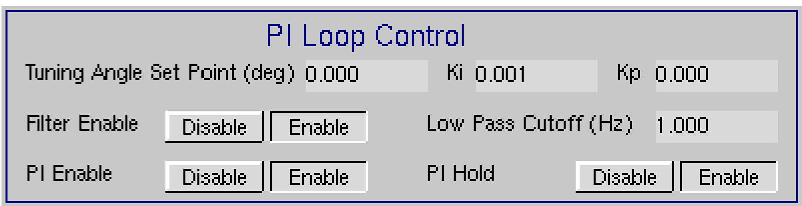
Observations:

1. Phase response is almost 0° up to 20Hz, this makes it ideal for proportional integral feedback control.
2. For higher frequencies of the phase response is very noisy. At the mechanical resonance modes, the phase changes by 180°. This is what makes compensation tough!

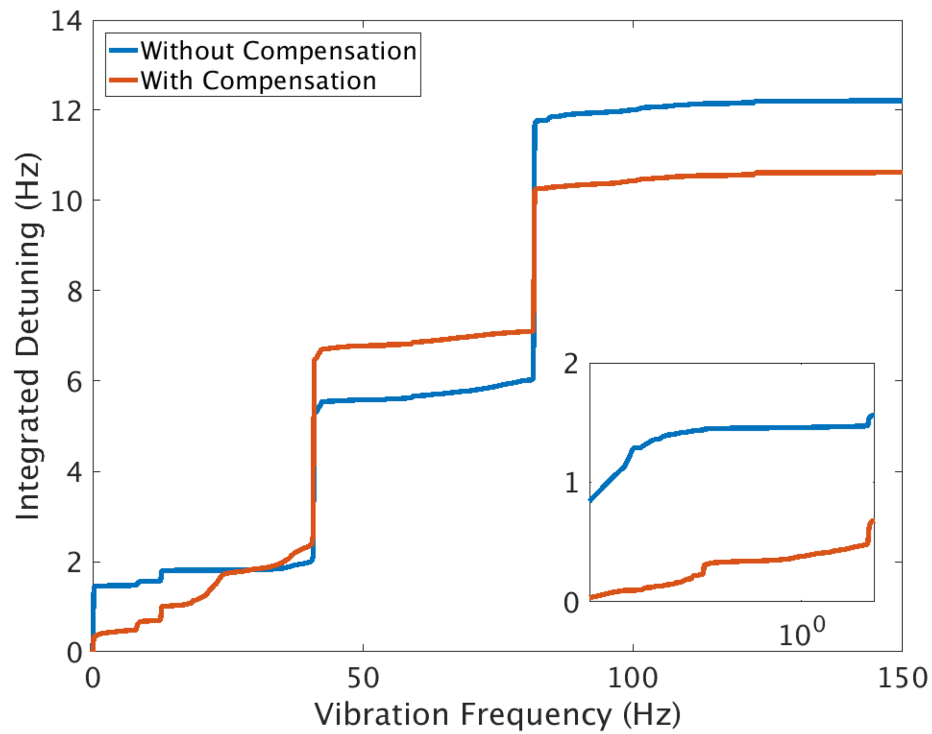
Feedback Compensation: Low Frequency

Proportional Integral Feedback Loop

1. First use a low pass IIR filter (2nd order Butterworth) to attenuate high frequency components.
2. Apply proportional integral control on the low passed signal.



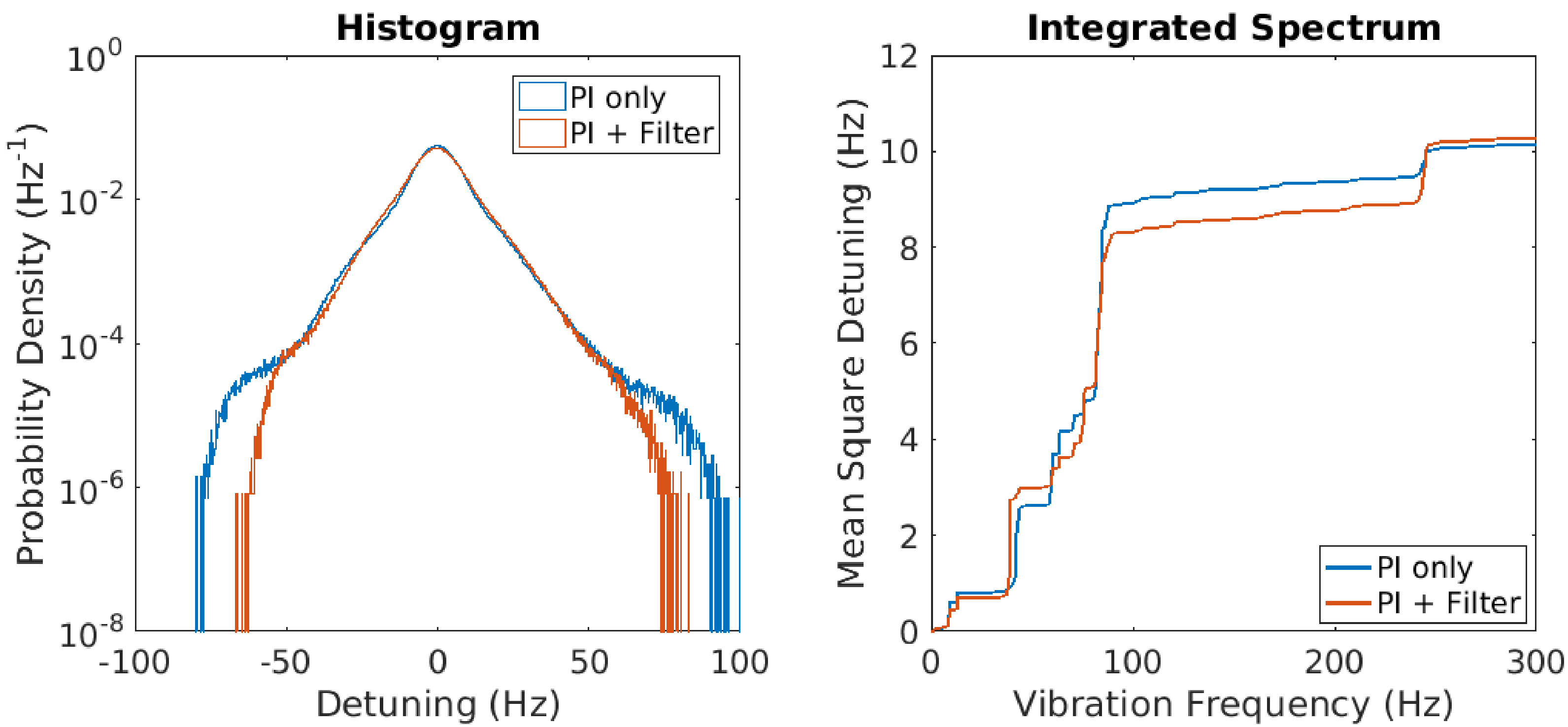
Very effective for sub-Hz frequencies!



Feedback Compensation: High Frequency

Band-pass filter Feedback loop

Use band-pass filters coupled to phase shifters to compensate for specific frequency components.

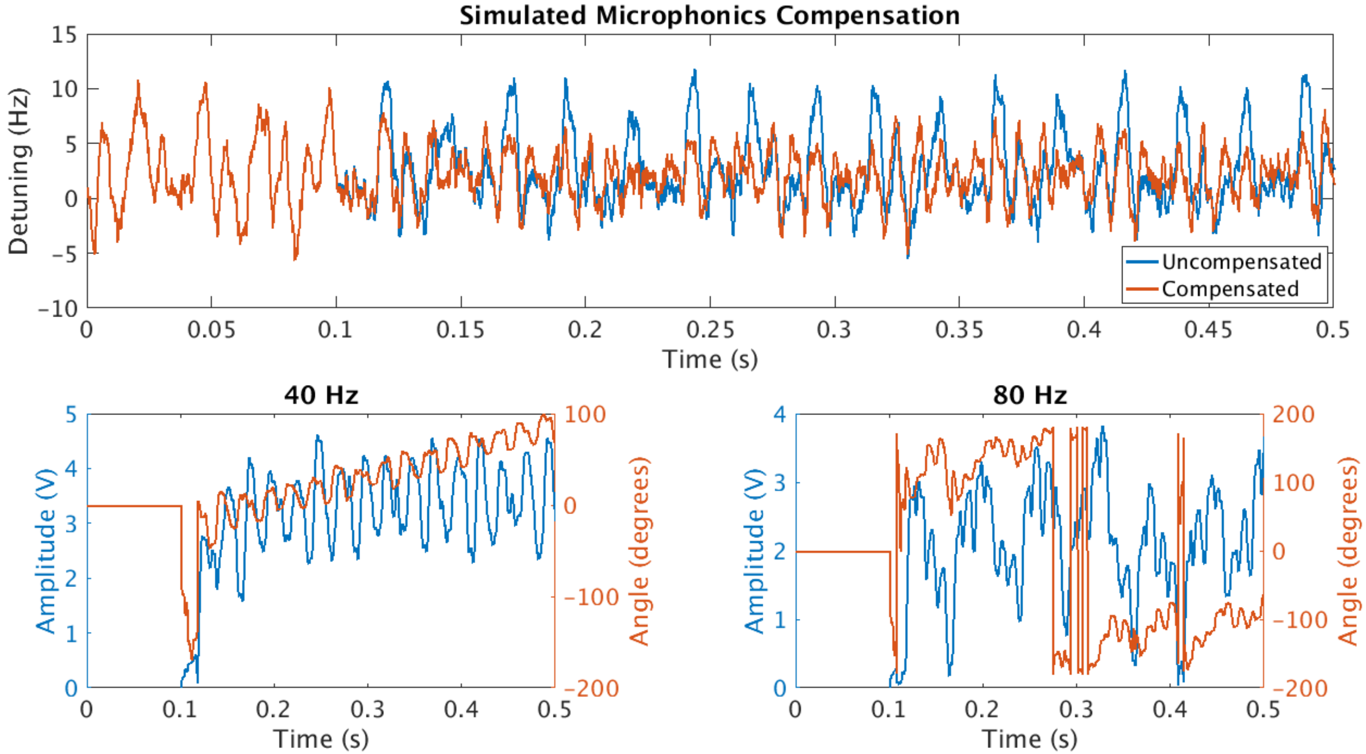


Three parameters have to be adjusted for each frequency. Getting to the optimal settings is not easy.

Not successful in first try!

Frequency domain Least Mean Square

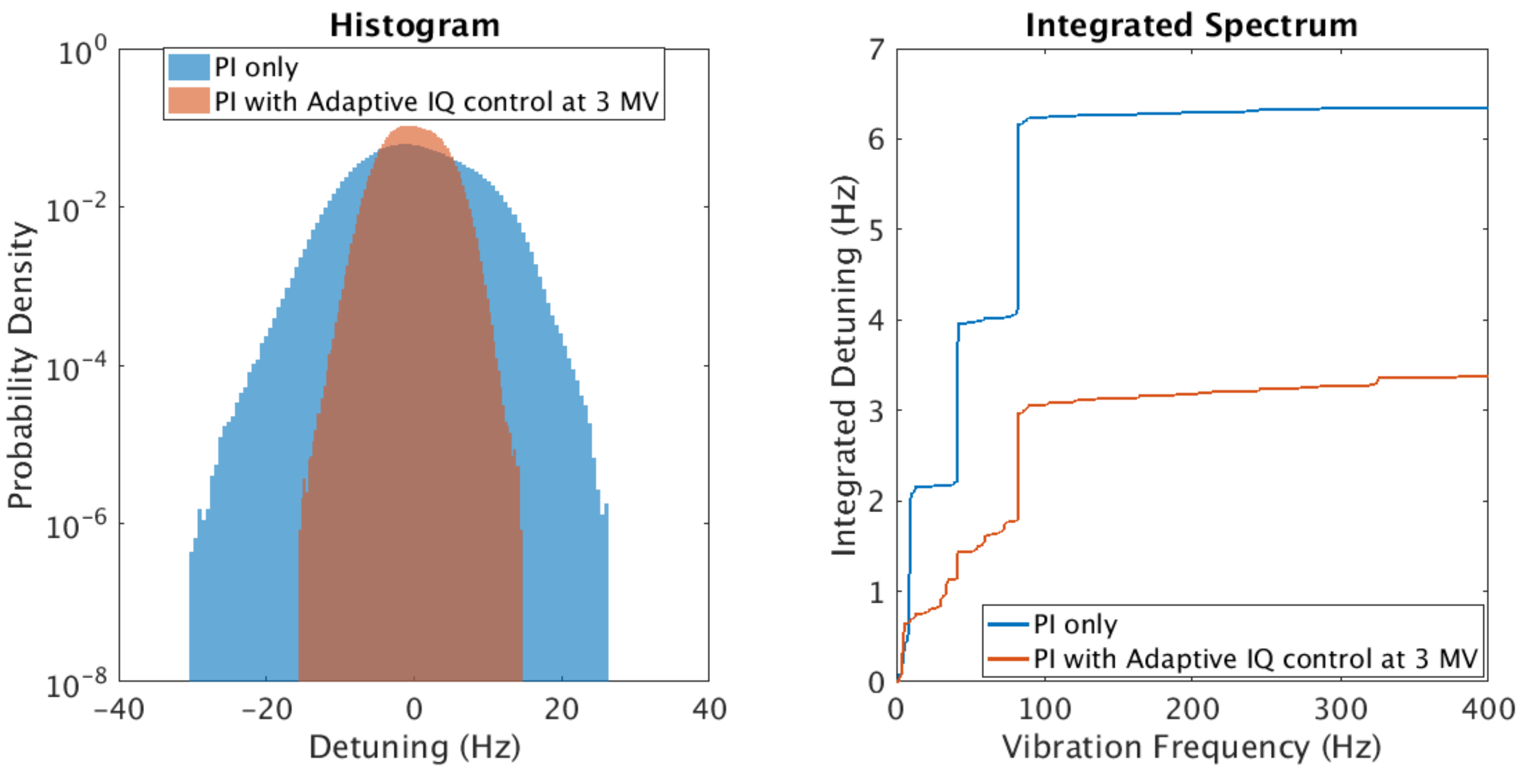
- Most of the vibration energy is concentrated near several frequencies.
- The compensation system must put out sinusoidal signals at these frequencies at the correct amplitude and phase so that the microphonics detuning due to the vibrations get cancelled.
- The Least Mean Square (LMS) algorithm adjusts the amplitude and phase of these signals in order to minimize the mean square of detuning.



Algorithm requires three parameters: ω_m, μ_m, φ_m

Results: Cavity 6 (Stiffened)

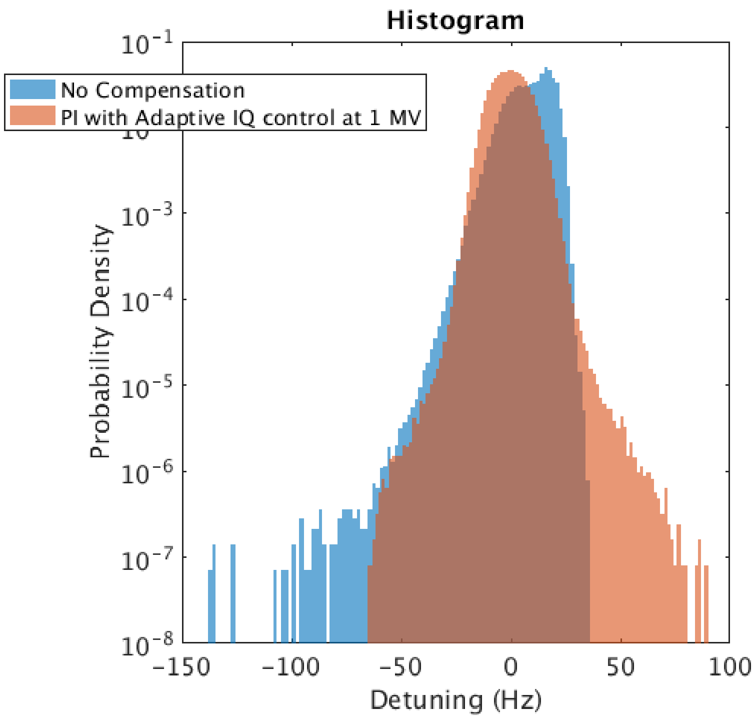
The algorithm was applied on three frequencies: 8 Hz, 40 Hz and 80Hz.
The parameters were adjusted manually by looking at the detuning signal.

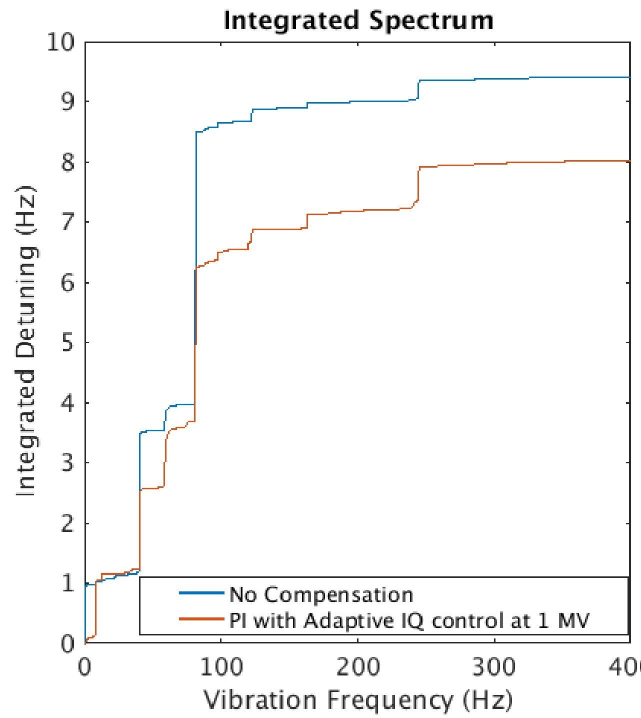


Algorithm is stable! Reduced peak detuning from 30.2 Hz to 15.5 Hz.

Results: Cavity 3 (Unstiffened)

The same algorithm was applied on the same frequencies as earlier.



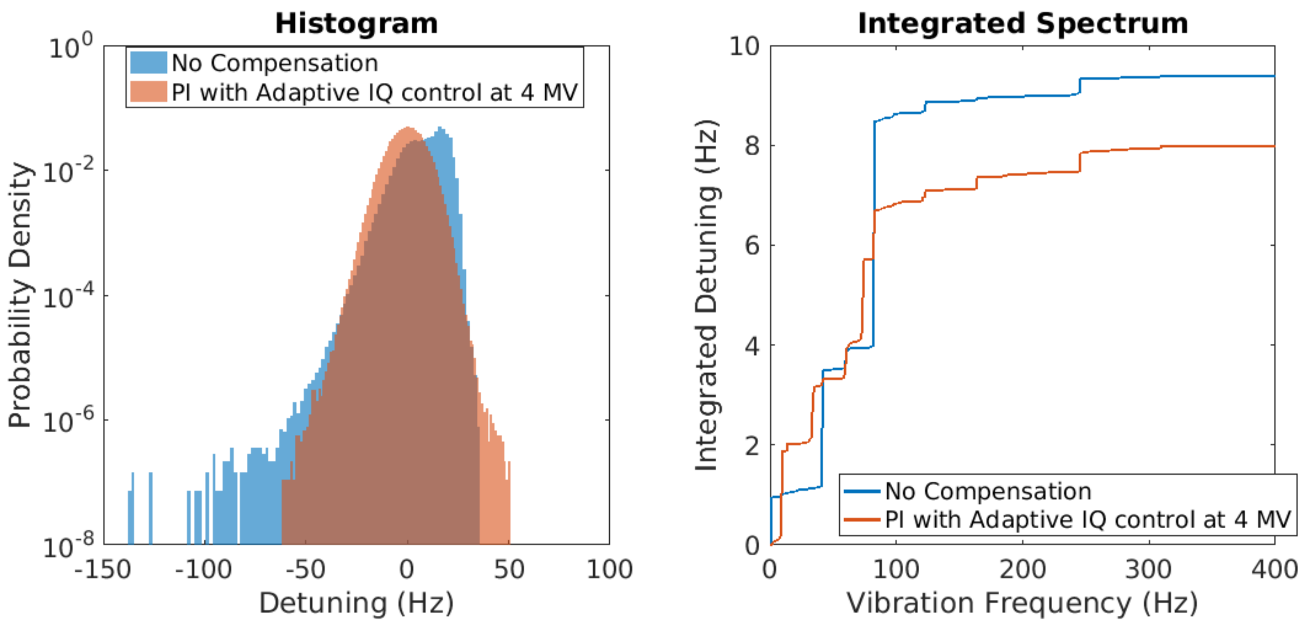


However, manual adjustment of the transfer function phase proved inadequate.

The phase φ_m must be determined automatically!

Results: Cavity 3 continued

The transfer function phase which was an user defined parameter must also be determined by the algorithm. Apply the LMS technique to get the phase.



- Peak detuning without compensation measured at 1MV: 137.4 Hz; with compensation measured at 4MV: 61 Hz
- In the modified algorithm, the estimated transfer function phase slowly oscillates around some optimal value.
- The oscillation becomes worse, if the Q of vibration peak is low and the algorithm becomes ineffective.

Summary

- The main linac cavites used for energy recovery in the CBETA project are powered by solid state power amplifiers capable of delivering up to 5kW which restricts the maximum allowable detuning to 54 Hz in order to sustain a cavity voltage of 6 MV.
- Microphonics detuning measurements on the main Linac cavities indicate that while the peak detuning in stiffened cavities is well below the limit of 54 Hz, microphonics the un-stiffened cavities (>130 Hz) needs to be reduced.
- Low frequency microphonics (sub Hz) can be effectively compensated by using proportional integral feedback control on the detuning signal processed by a low pass filter.
- The frequency domain Least Mean Square technique compensates for detuning by controlling the amplitude and phase of the actuator signal using gradient descent on mean square detuning.
- The algorithm was applied successfully to a both a stiffened cavity (33 Hz to 15.5 Hz) and an unstiffened one (137 Hz to 61 Hz).
- More work needs to be done to improve transfer function phase estimation.

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People who contributed to this work.

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5. Peter Quigley
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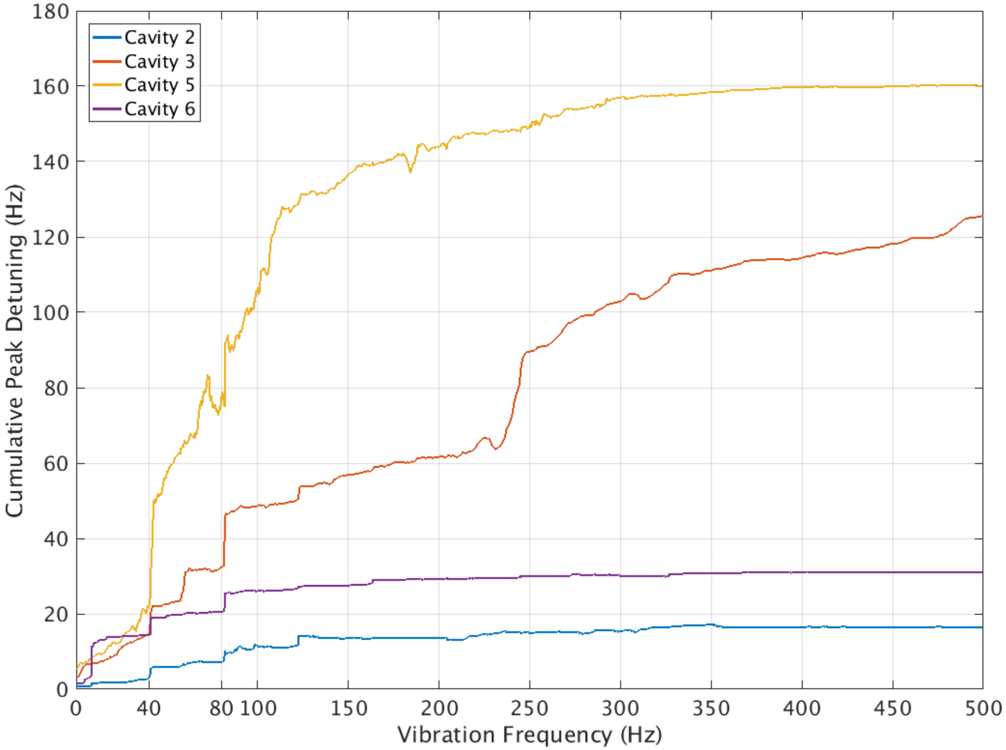
Thank you!

Appendix 1: Peak Detuning Algorithm

Algorithm

1. Find the Discrete Fourier Transform the whole signal.
2. Make all coefficients of the Discrete Fourier Transform zero which are greater than some frequency f.
3. Do the inverse Fourier transform of this modified DFT and find the absolute max value of this signal.
4. Repeat for many frequencies.



Appendix 2: Filter Expressions

1. Use a low pass filter with an integrator to take care of low frequencies. Something similar is already done in the current version of the control system.

$$H_{\mathrm{LPI}} = \frac{K_i \omega_c}{s\left[\left(\frac{s}{\omega_c}\right)^2 + \frac{\sqrt{2}s}{\omega_c} + 1\right]^n}$$

ω_c is the -3n dB cutoff frequency.
n is the filter order, higher order corresponds to a steeper roll off.
K_i is the strength of the integral term of the feedback loop.

2. Use band pass filters to suppress specific frequency components.

$$H_{BP} = \left[\frac{K_b\left(\frac{s}{\omega_c} + P_b\right)}{\left(\frac{s}{\omega_c}\right)^2 + \frac{2s}{\omega_c Q} + 1}\right]^n$$

ω_c is the center frequency.
n is the filter order, higher order corresponds to a steeper roll off.
P_b is set to 0.
Q controls the width of the filter.
K_b is the strength of the band-pass term of the feedback loop.

Appendix 2 Contd: Phase Shifter

To shift a signal of frequency f_c by a phase φ_c, we can just shift the signal by some number of samples given by: $N = \frac{f_s \phi_c}{2\pi f_c}$

For f_c = 8 Hz and φ_c = 150°, the total buffer length is N = 102. Allocating a buffer of this size for each of the 10 frequency channels requires more memory than what we have left. We have two options:

1. Switch to a different memory allocation scheme. However, we will still be limited by the maximum buffer length available.
2. Use a 4 tap FIR filter, i.e. use four coefficients. Then the filter response is given by:

$$H(f) = \sum_{i=1}^{4} a_{n_i} \mathrm{e}^{i n_i \frac{2\pi f}{f_s}}$$

Constrain the filter with the conditions:

$$H(f_c) = \mathrm{e}^{i\phi_c} \qquad \left.\frac{\mathrm{d}H}{\mathrm{d}f}\right|_{f=f_c} = 0$$

The constraints give us 4 constraints, to solve for the a_{ni}. We still have the freedom to choose the placement of the filter taps n_i. So we can optimize for least memory use.

Appendix 3: Frequency domain LMS

Idea: Treat this as a real time least squares optimization problem.

Actuator Signal: $$\boxed{u_{pz}(t_n) = \sum_m I_{mn}\cos\omega_m t_n + Q_{mn}\sin\omega_m t_n}$$

Assuming that the tuner is a LTI system the detuning can be written as:

$$\delta f_{\text{comp}}(t_n) = \delta f_{\text{ext}}(t_n) + \sum_m \alpha_m \{I_{mn}\cos(\omega_m t_n - \phi_m) + Q_{mn}\sin(\omega_m t_n - \phi_m)\}$$

Where, α_m and φ_m describe the transfer function from actuator to detuning at the frequency ω_m.

Mean Square Detuning: $F_n = E[\{\delta f_{\text{comp}}(t_n)\}^2] \sim \{\delta f_{\text{comp}}(t_n)\}^2$

Minimize the mean square detuning using a gradient descent algorithm.

$$\boxed{\begin{aligned} I_{m,n+1} &= I_{m,n} - \mu_m \delta f_{\text{comp}}(t_n)\cos(\omega_m t_n - \phi_m) \\ Q_{m,n+1} &= Q_{m,n} - \mu_m \delta f_{\text{comp}}(t_n)\sin(\omega_m t_n - \phi_m) \end{aligned}}$$

ω_m ,μ_m and φ_m are fixed by the operator.

$$\boxed{\phi_{m,n+1} = \phi_{m,n} - \eta_m \delta f_{\text{comp}}(t_n)\{I_{m,n+1}\sin(\omega_m t_n - \phi_m) - Q_{m,n+1}\cos(\omega_m t_n - \phi_m)\}}$$

Appendix 4: LMS as linear feedback

The basic LMS algorithm is summarized in the following equations:

$$I_{m,n+1} = I_{m,n} - \mu_m \delta f_{\mathrm{comp}}(t_n)\cos(\omega_m t_n - \phi_m)$$
$$Q_{m,n+1} = Q_{m,n} - \mu_m \delta f_{\mathrm{comp}}(t_n)\sin(\omega_m t_n - \phi_m)$$
$$u_{pz}(t_n) = \sum_m I_{mn}\cos\omega_m t_n + Q_{mn}\sin\omega_m t_n$$

Now assuming (1) Method is continuous. (2) There is no phase lag involved.

$$u_{pz}(t) = -\sum_m \tilde{\mu}_m \mathrm{Re}\left[\mathrm{e}^{-\mathrm{i}\omega_\mathrm{m}\mathrm{t}}\int_0^\mathrm{t} \mathrm{e}(\mathrm{t}')\mathrm{e}^{\mathrm{i}\omega_\mathrm{m}\mathrm{t}}\mathrm{dt}'\right]$$

This is clearly Linear.

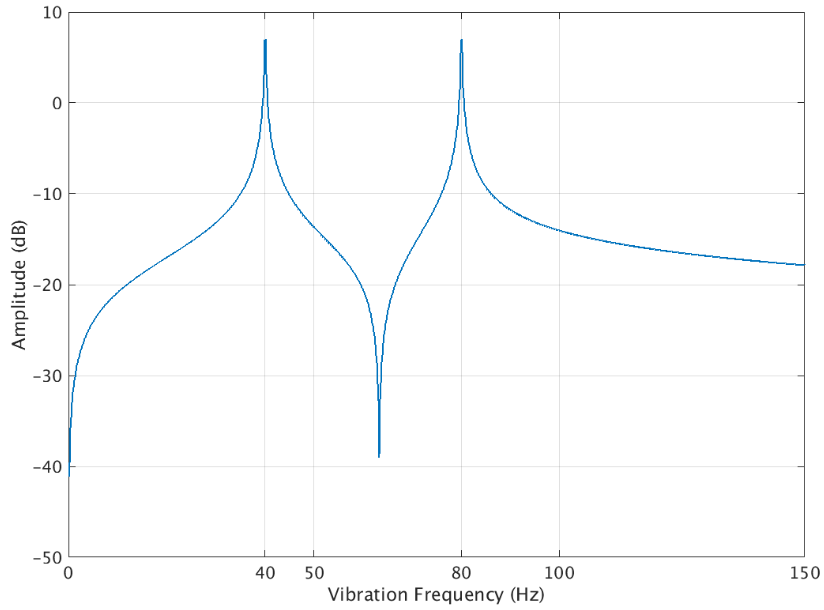
To understand how LMS works, let's see what it does to $\mathrm{e}(\mathrm{t}) = \mathrm{A_s}\cos(\omega_\mathrm{s}\mathrm{t} + \phi_\mathrm{s})$

$$u_{pz}(t) = -\boxed{\sum_m \tilde{\mu}_m \frac{\omega_s}{\omega_s^2 - \omega_m^2}} A_s[\sin(\omega_s t + \phi_s) - \sin(\phi_s)]$$

Appendix 4 Contd: Transfer Function

The response of the LMS controller can be written as:

$$|C(\omega_s)| = \sum_m \frac{\tilde{\mu}_m \omega_s}{\omega_s^2 - \omega_m^2}$$



The LMS algorithm is approximately equivalent to a set of band pass filters!

Douglas MacMartin, A feedback perspective on the LMS disturbance feedforward algorithm, American Control Conference, 1994, 1994, pp. 1632-1636 vol.2, doi: 10.1109/ACC.1994.752347