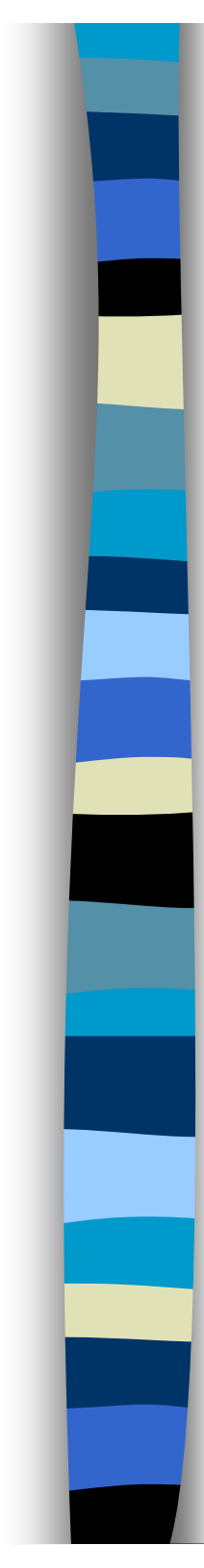


Understanding Acoustic Feedback



... a closer look into that annoying phenomena



I've heard feedback called
a lot of things before
but

Larsen ?

Another Name For Feedback is: The “Larsen Effect”



Named after the Danish physicist
Soren Larsen (1871-1957)

Some are quick to point out that not all
feedback in audio is bad...



Larsen Forever

A project by pouvoir d'achat'

www.artcontemporain.lu/larsen/larsen.html

Ways to Produce Larsen (acoustic feedback):

Quoted from Web Site

- “Increasing the available amplification is most evident condition to get satisfying feedback.”
- “Distance is the other factor. Moving the microphone (or speaker) to shorten the acoustic path to the loudspeaker can often increase feedback.”
- “Boosting tone controls indiscriminately may help.”
- “Room surfaces that are hard and reflective such as glass, marble, wood, may enhance reverberation and by this feedback.”
- “More open microphones (or other transducer-devices) increase feedback.”

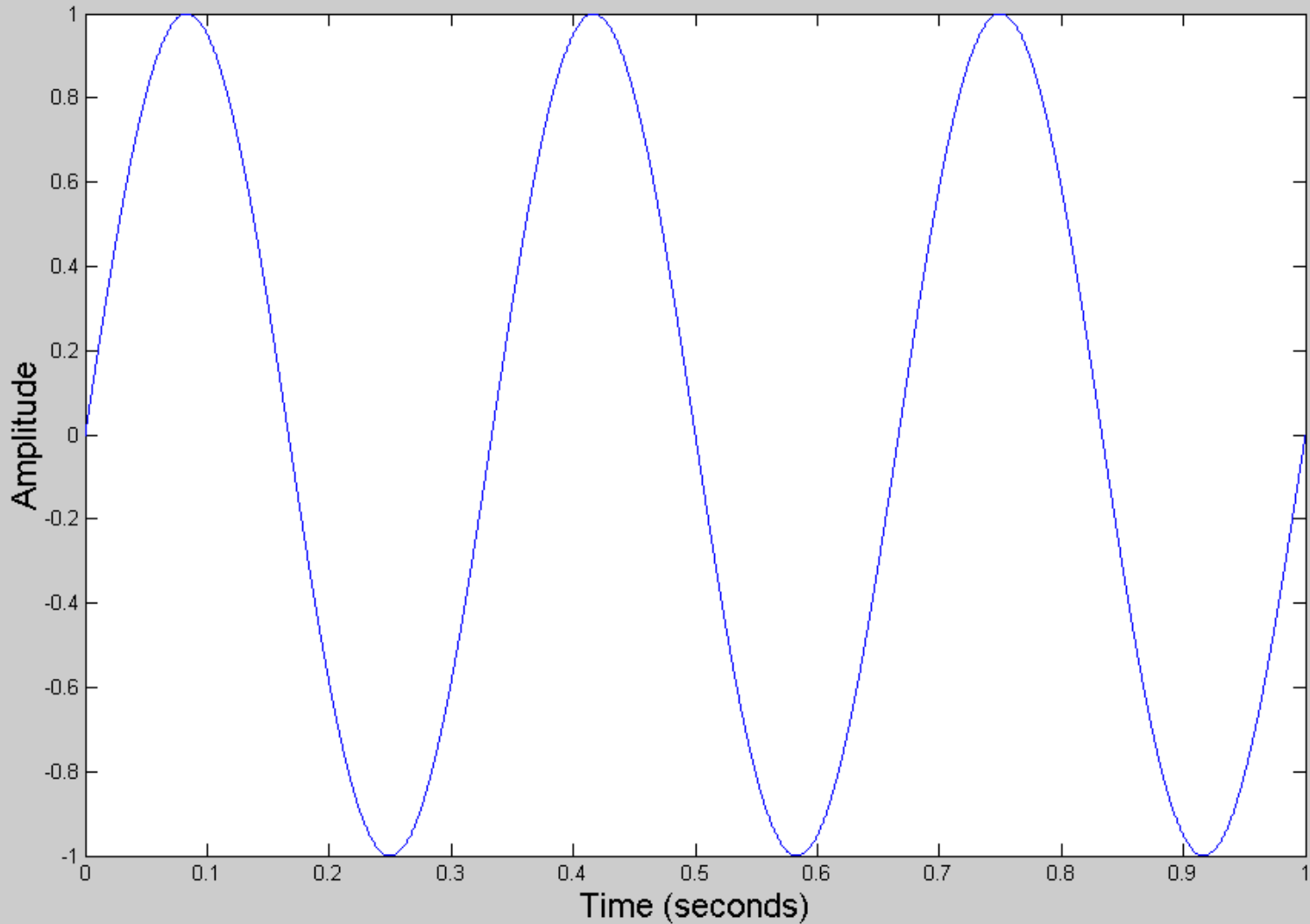


But First ...

Terms & Concepts

Sinusoid: $A\sin(\omega t + \emptyset)$

A =Peak Amplitude; $\omega=2\pi f$ Angular Frequency;
 f =frequency; Units are Cycles/Second or Hertz (Hz)
 \emptyset =Phase (Radians); Degrees=Radians* $180/\pi$

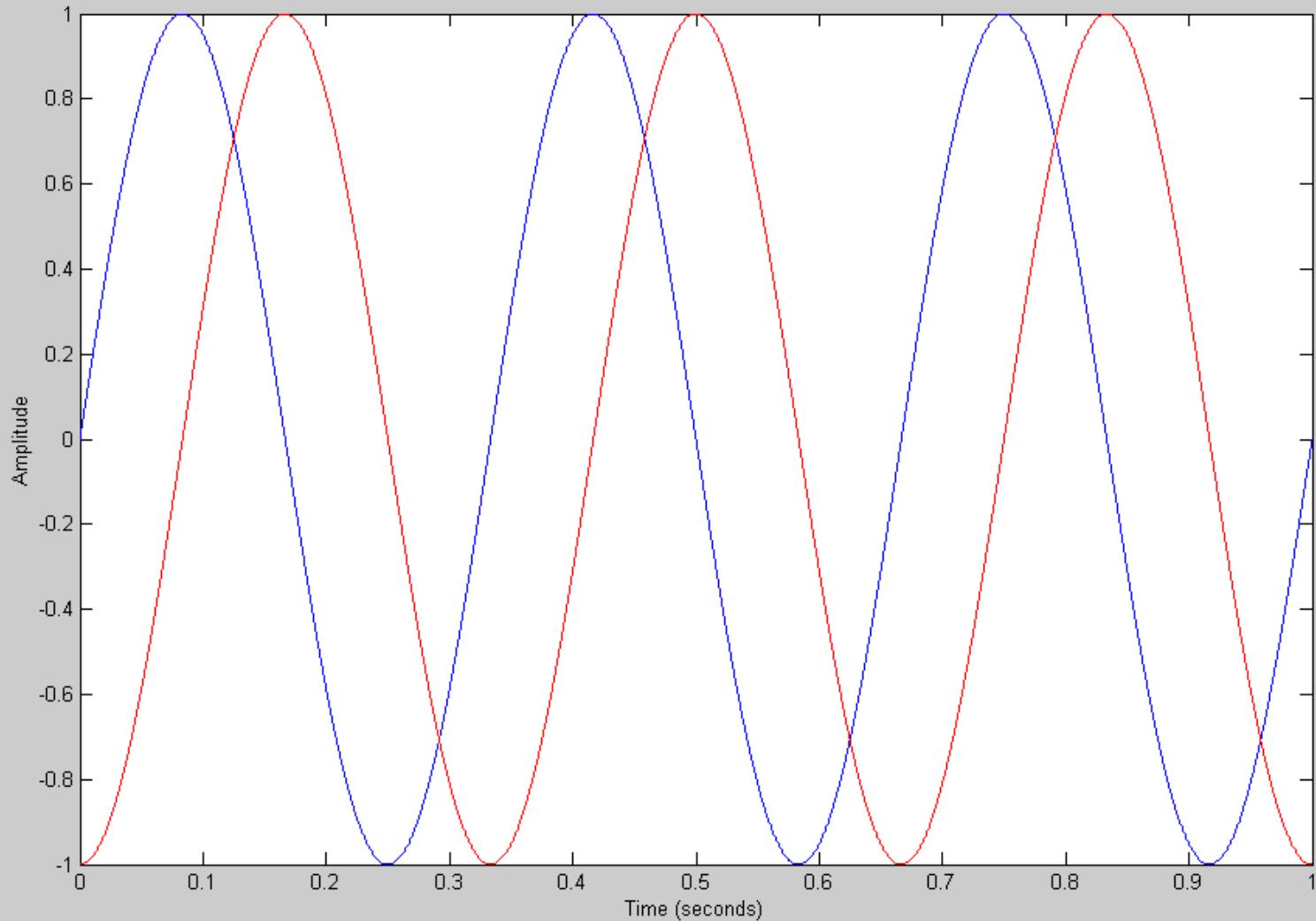


Terms:

Magnitude: (Think Peak Amplitude)

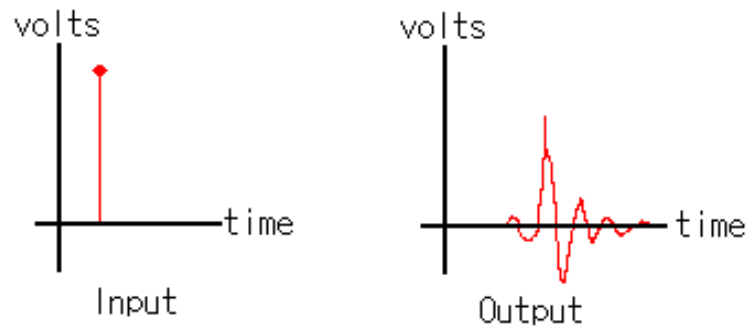
Unity Gain: (Think Multiply by 1)

Phase: (Red curve's phase is shifted 90 degrees from Blue)



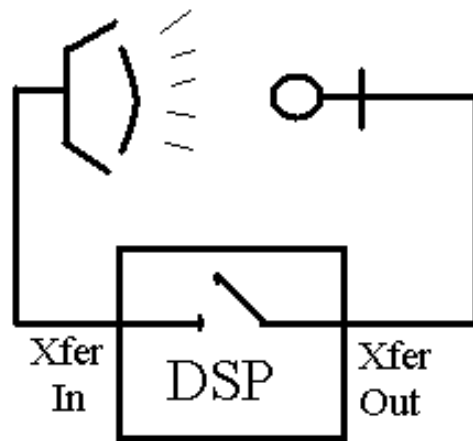
For added insight we'll use the

Transfer Function



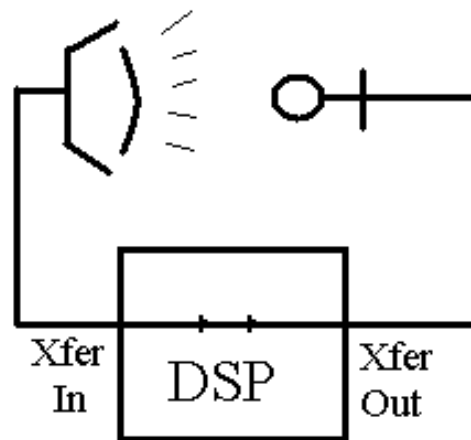
$$\text{Transfer Function} = \frac{\text{Fourier Transform[Output]}}{\text{Fourier Transform[Input]}}$$

Open Loop Response



No Feedback Between
Speaker and Microphone

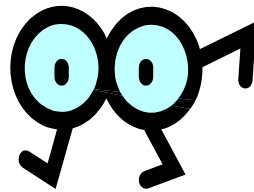
Closed Loop Response



Feedback Between
Speaker and Microphone

What affects feedback?

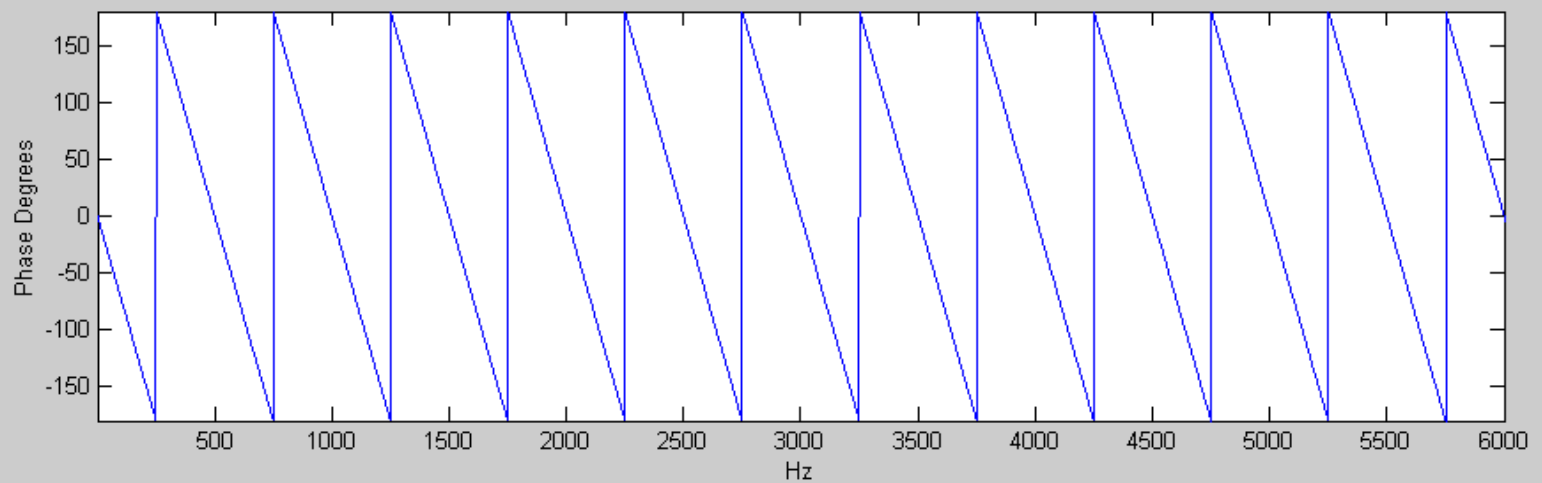
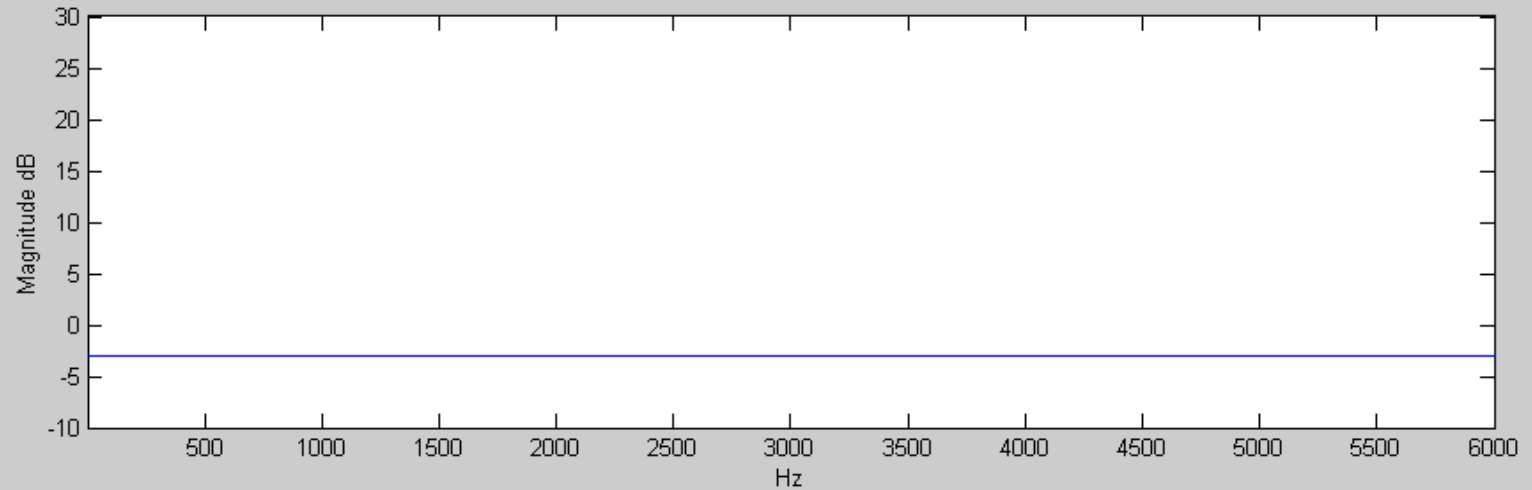
Transfer function characteristics of a delay block



Magnitude is flat and the phase is linear (for linear plot). Phase decreases linearly as the frequency increases. From any two points on the phase plot you can calculate the slope of the phase and determine the delay.

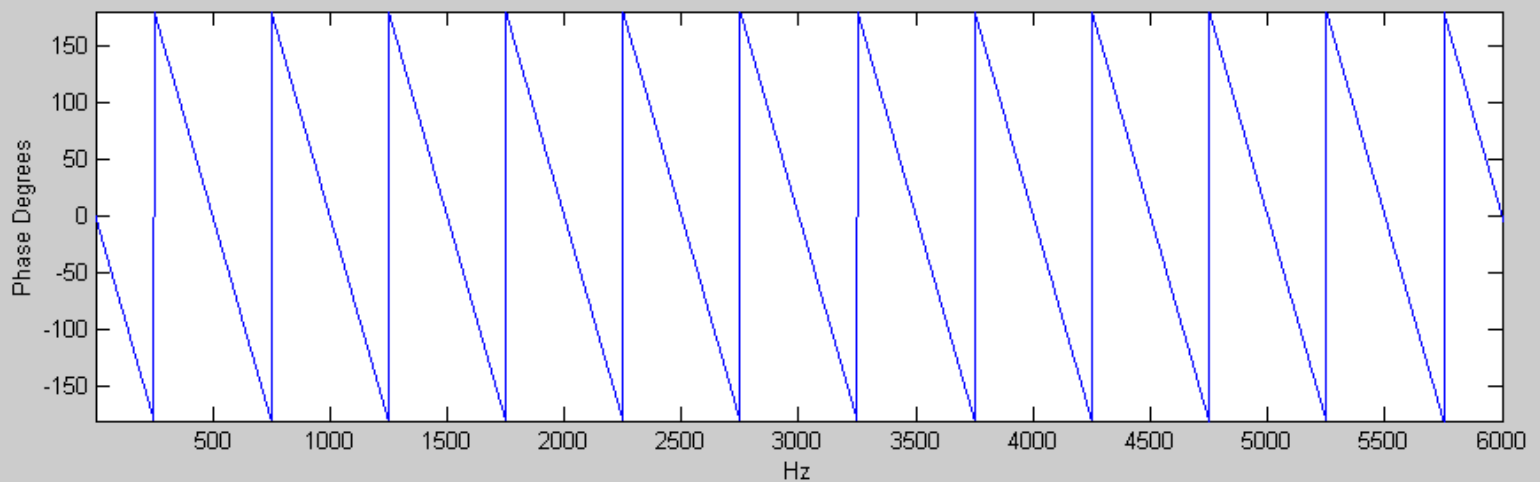
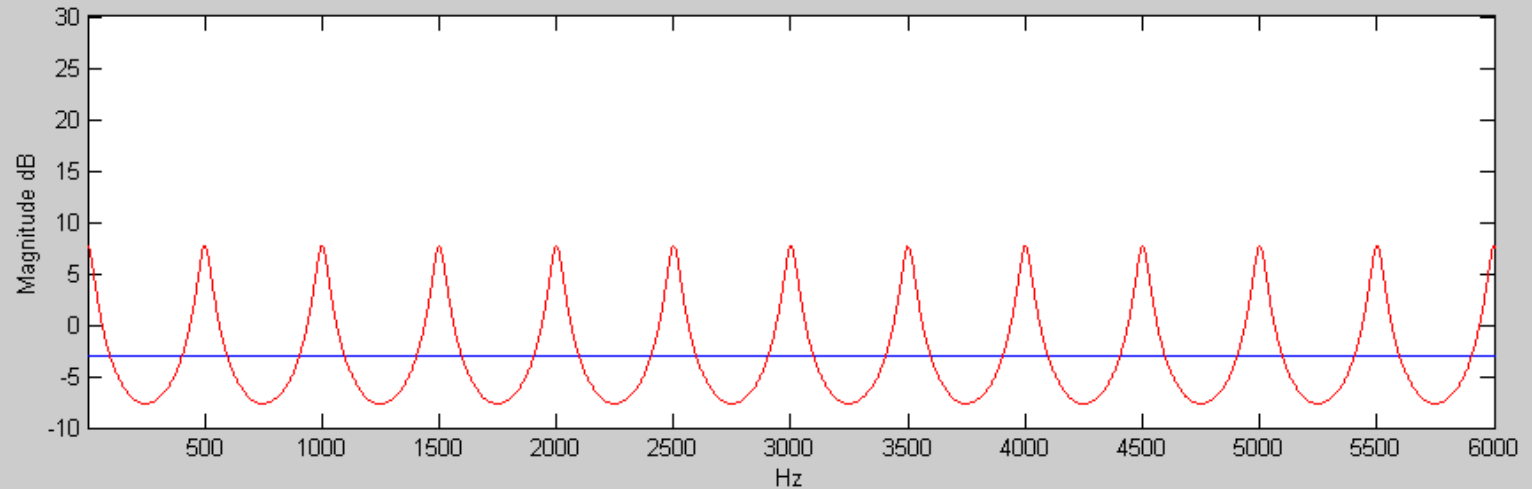
Transfer Function

Delay = 2.0 ms, Gain = -3 dB



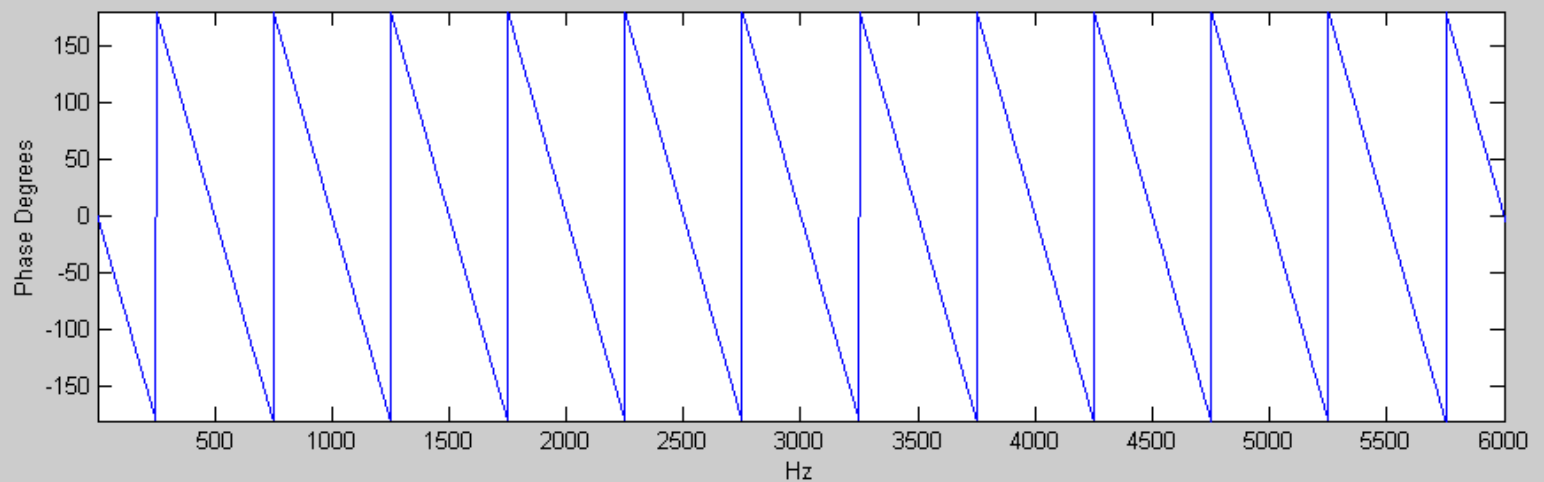
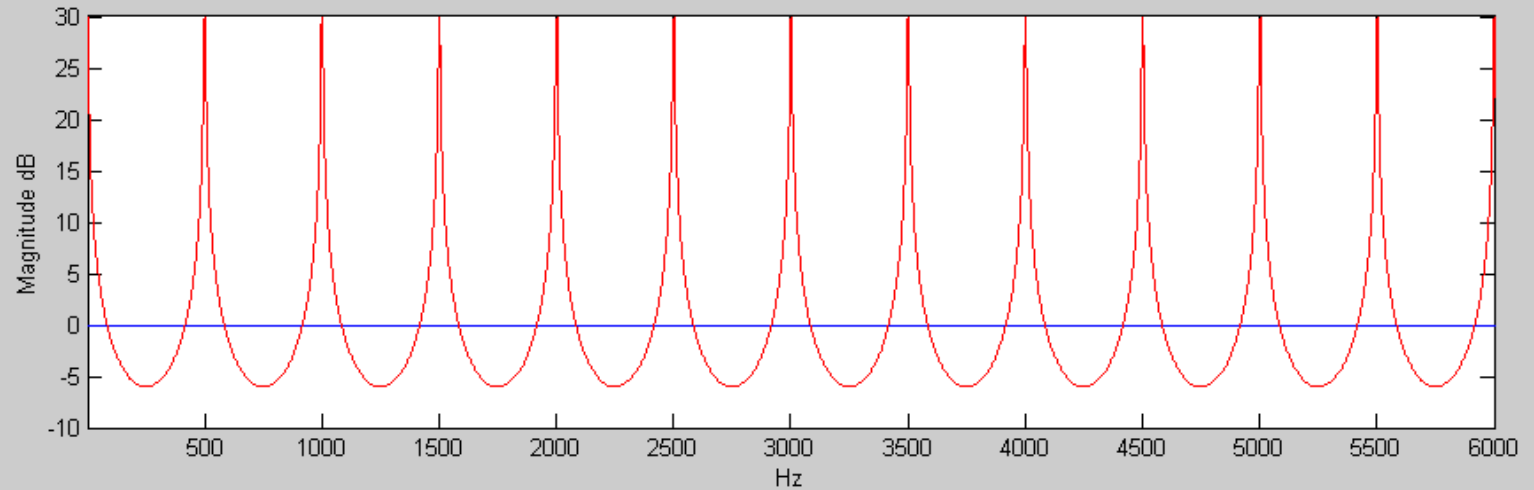
Add Closed Loop Response

Delay = 2.0 ms, Gain = -3 dB



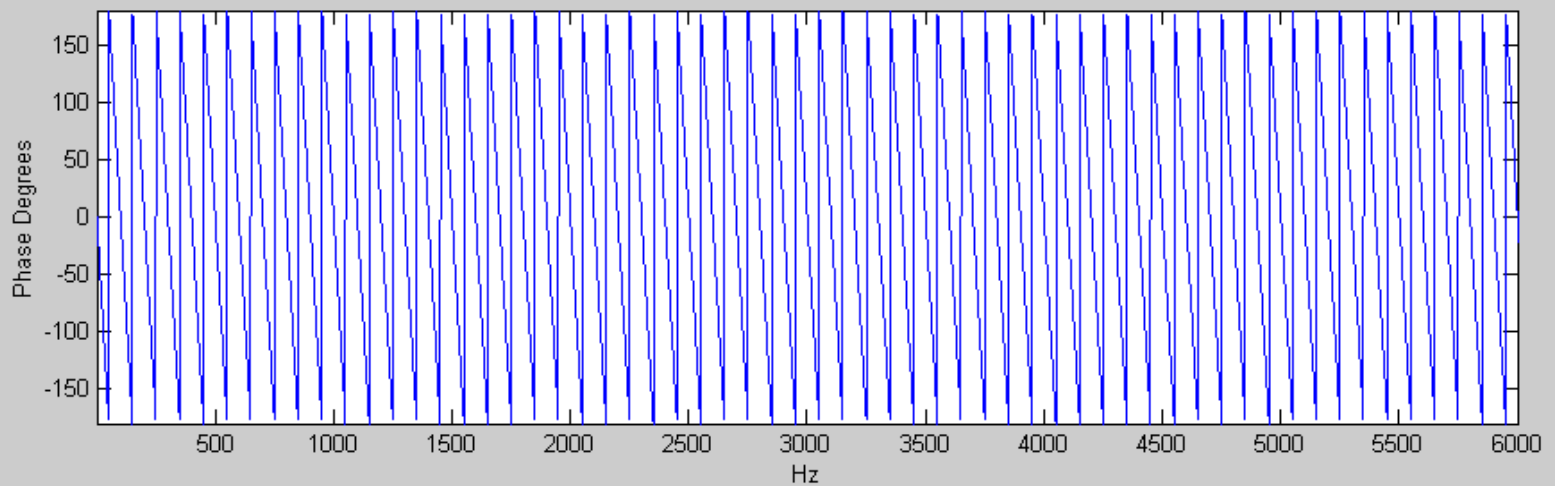
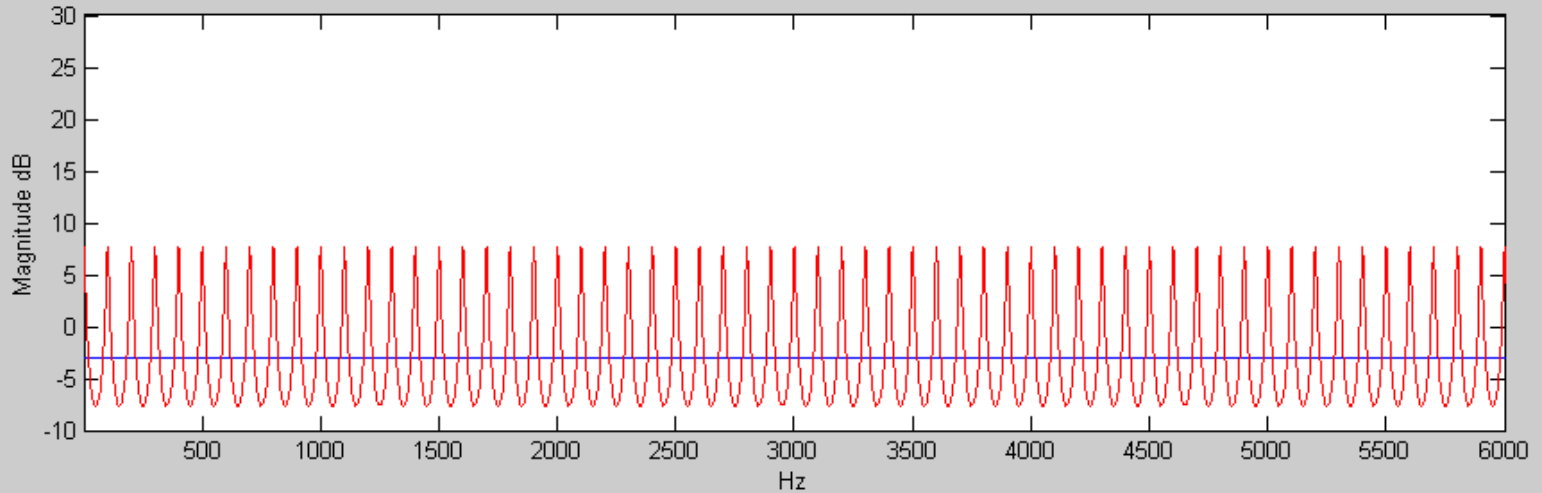
Open/Closed Loop Responses

Delay = 2.0 ms, Gain = 0 dB



Open/Closed Loop Responses

Delay = 10.0 ms, Gain = -3 dB





Feedback Characteristics

- Increasing system delay, increases the number and reduces the spacing, of potential feedback frequencies.

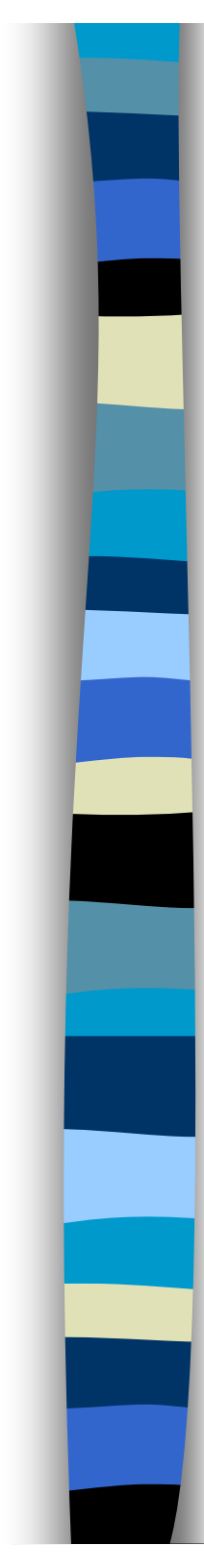
$$\text{potential feedback frequency spacing} = 1/\text{delay}$$

$$1/0.002 \text{ sec.} = 500 \text{ Hz spacing}$$

$$1/0.010 \text{ sec.} = 100 \text{ Hz spacing}$$

$$1/0.1 \text{ sec.} = 10 \text{ Hz spacing}$$

Increasing delay increases the number of potential feedback frequencies



Doesn't this conflict with the notion that decreasing delay increases feedback?

Can't forget that delay affects the rate of increase/decrease of a feedback component.

Open loop Gain dB
(above unity)

————— = Rate increase dB/sec
Delay sec



Feedback Characteristics

- Increasing system delay, increases the number and reduces the spacing, of potential feedback frequencies.
- System delay affects the rate at which feedback will grow and decay
- To bring a feedback frequency back into control, the open loop gain simply needs to be reduced below unity. However, dropping the gain just below unity can cause a very slow decay.

$$+1 \text{ dB}/0.010 \text{ sec} = 100 \text{ dB/sec}$$

$$+3 \text{ dB}/0.010 \text{ sec} = 300 \text{ dB/sec}$$

$$+1 \text{ dB}/0.100 \text{ sec} = 10 \text{ dB/sec}$$

$$+3 \text{ dB}/0.100 \text{ sec} = 30 \text{ dB/sec}$$

Same holds true for decay.



How does temperature affect feedback?

Temperature changes the speed of sound, which changes the effective delay between the speaker and microphone.

The speed of sound in air increases as temperature increases, reducing the delay. As delay is reduced, potential feedback frequencies shift higher.

Therefore as the temperature increases feedback frequencies shift higher.

From 71° F to 91° F (1129.7 ft/sec to 1150.8 ft/sec)

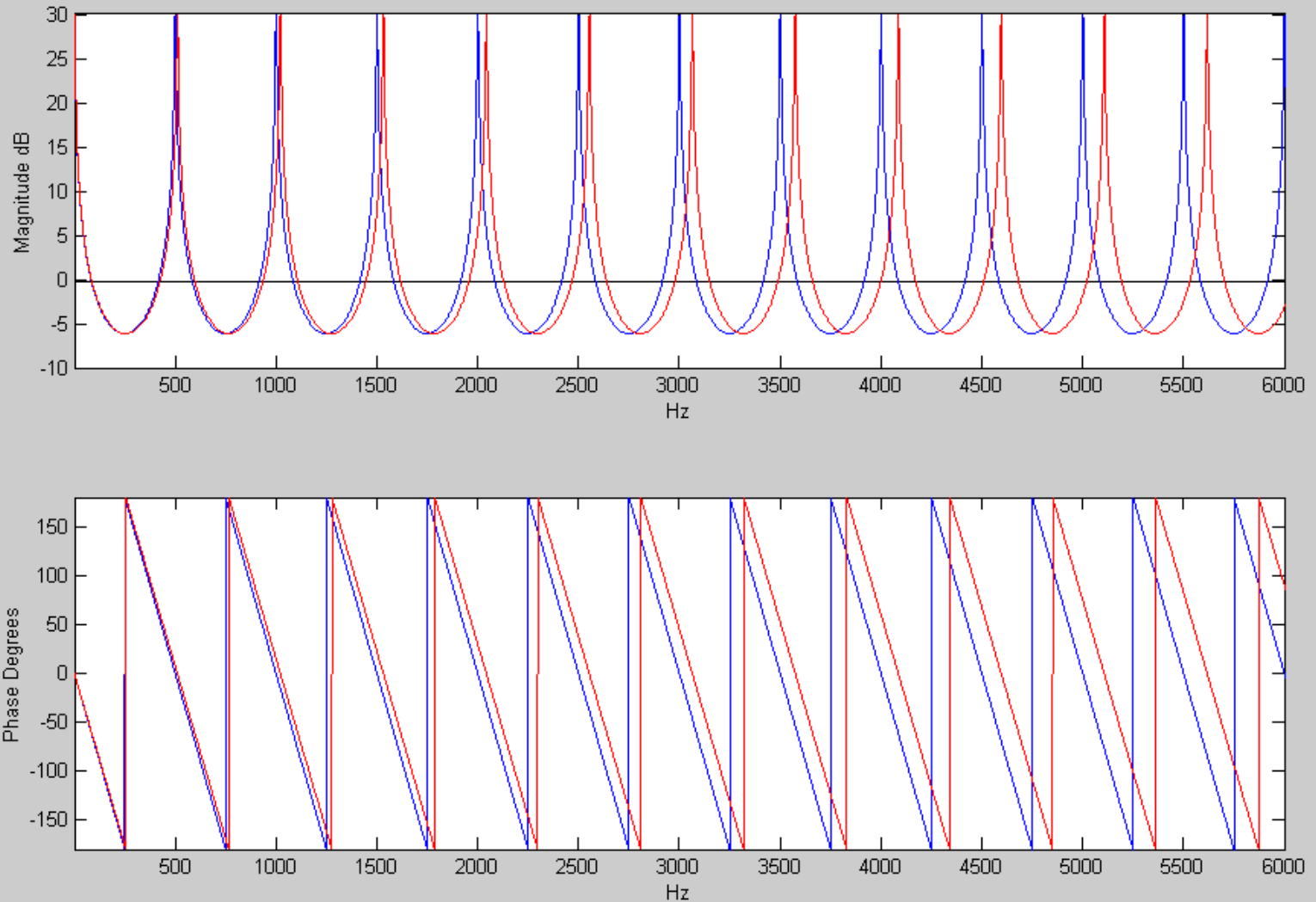
10 ft delay decreases by 0.16 ms

50 ft delay decreases by 0.81 ms

100 ft delay decreases by 1.62 ms

Feedback frequencies for 2.3 ft (2 ms) of delay at 71° F & 91° F

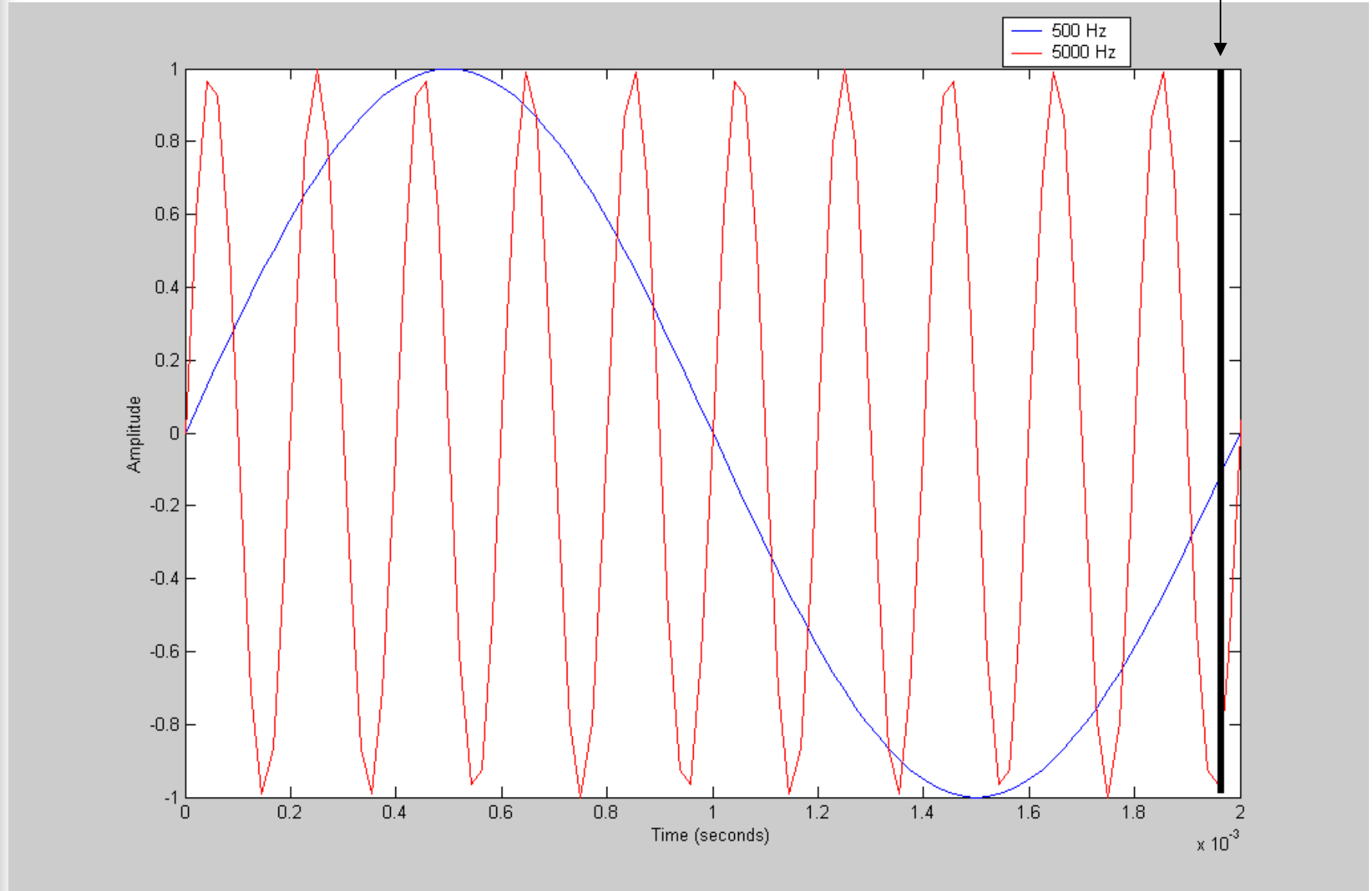
Higher feedback frequencies are shifted more than lower feedback frequencies



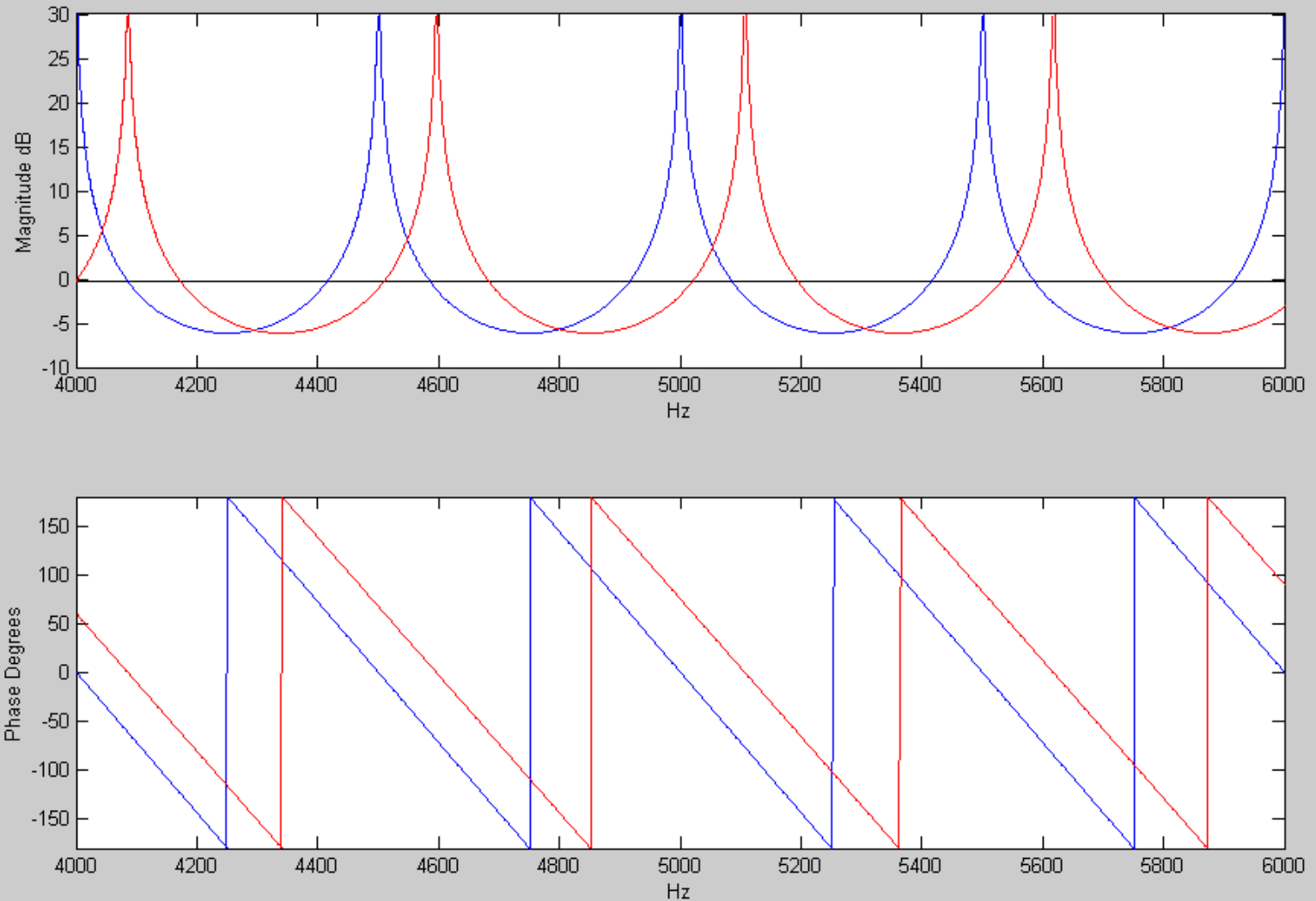
Why are higher frequencies affected more than lower frequencies?

2 ms of delay is changed 38 μ s due to a 20° temperature change. That represents 6 degrees of phase in the 500 Hz signal and 60 degrees of phase in the 5000 Hz signal.

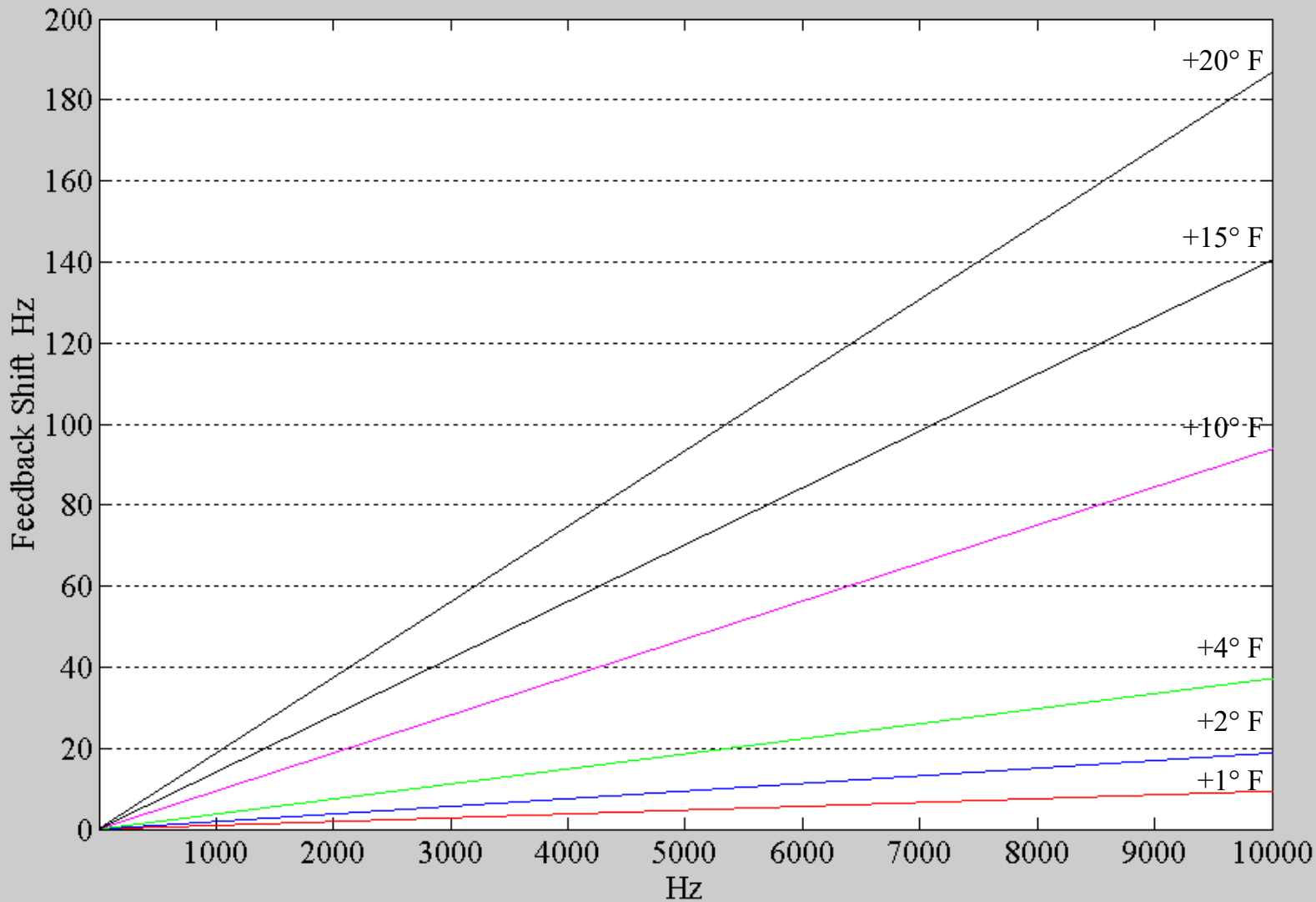
Delay change
due to temperature



Feedback frequencies for 2 ms of delay at 71° F & 91° F (zoom)



Feedback Frequency Shift For Various Temperature Changes (based on corresponding changes in delay)

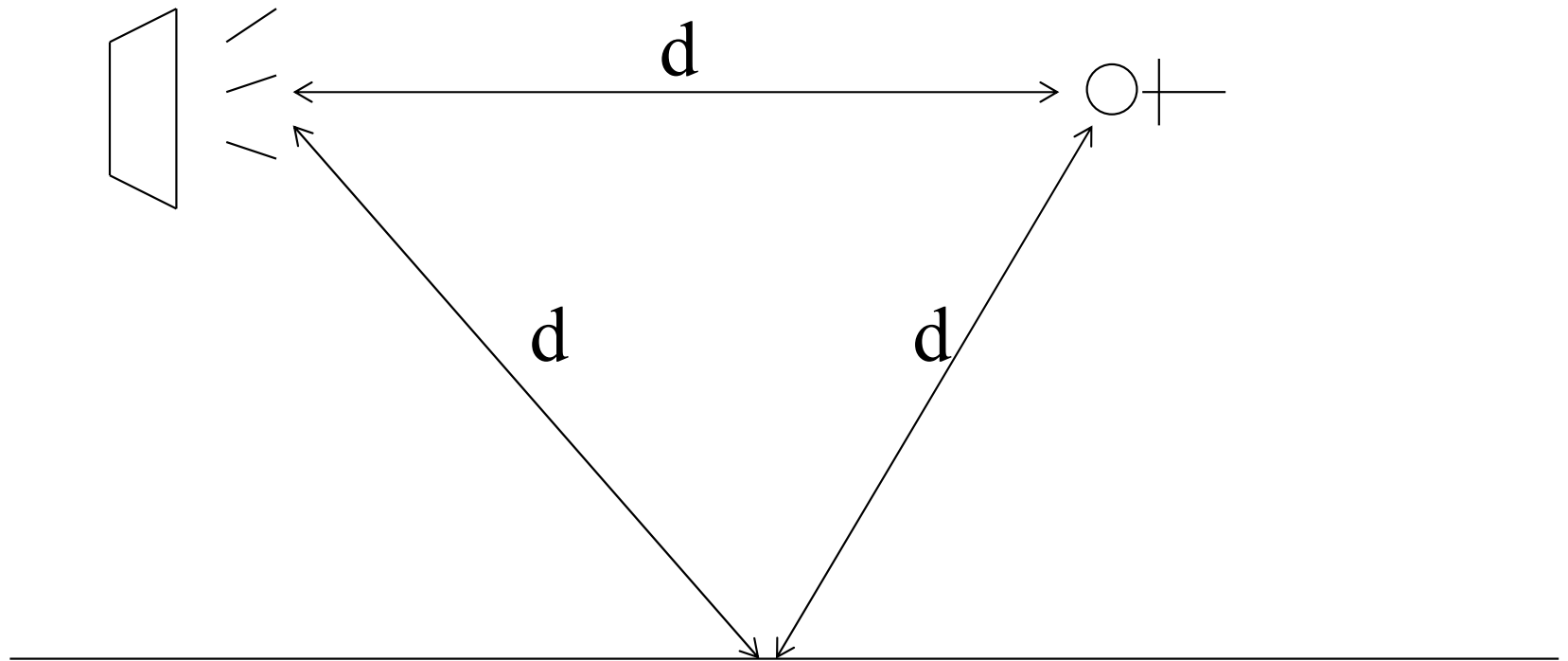




Feedback Characteristics

- Increasing system delay, increases the number and reduces the spacing, of potential feedback frequencies.
- System delay affects the rate at which feedback will grow and decay
- To bring a feedback frequency back into control, the open loop gain simply needs to be reduced below unity. However, dropping the gain just below unity can cause a very slow decay.
- As temperature increases, potential feedback frequencies shift higher. (the higher the frequency, the greater the shift)

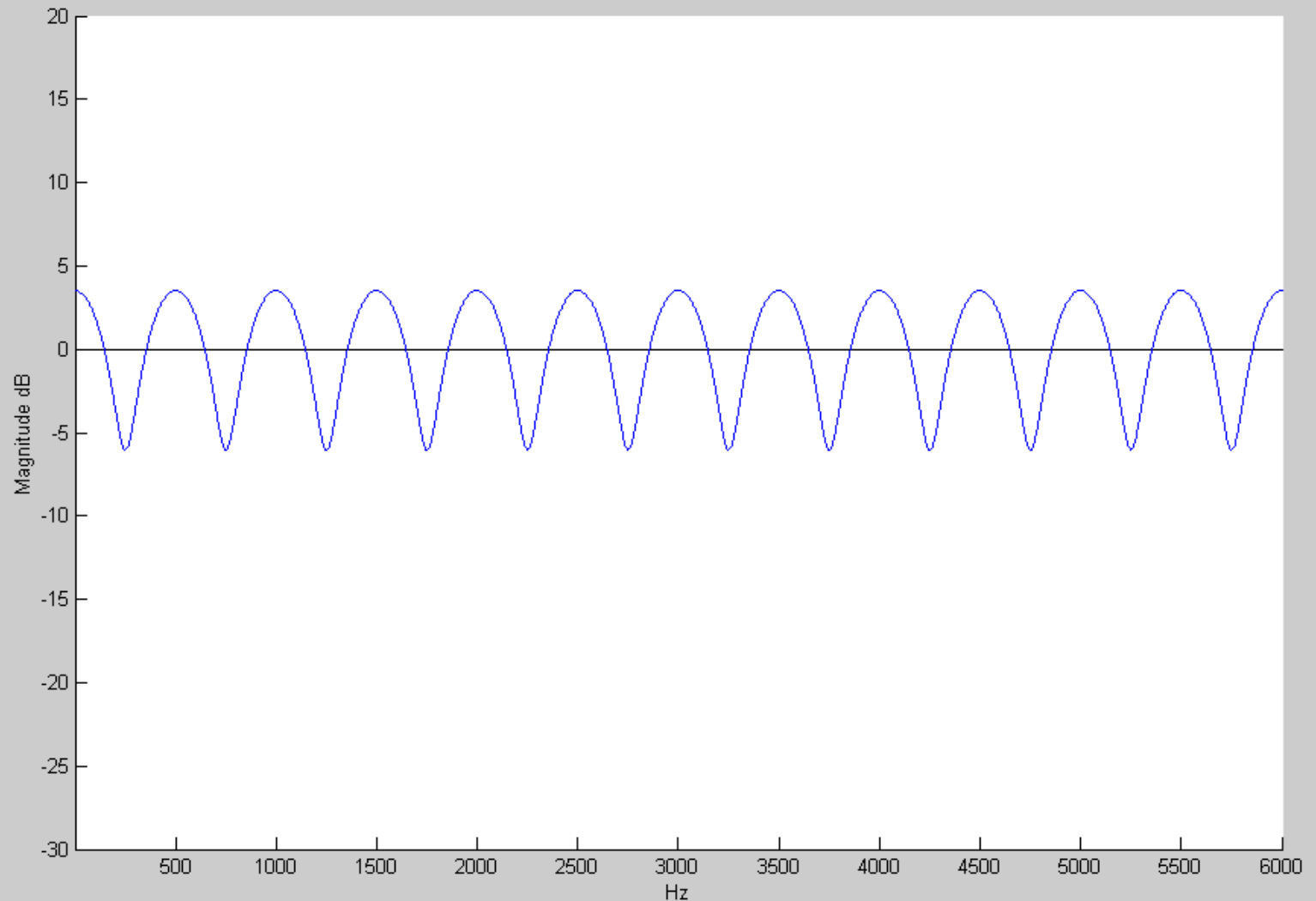
How do reflections affect feedback?



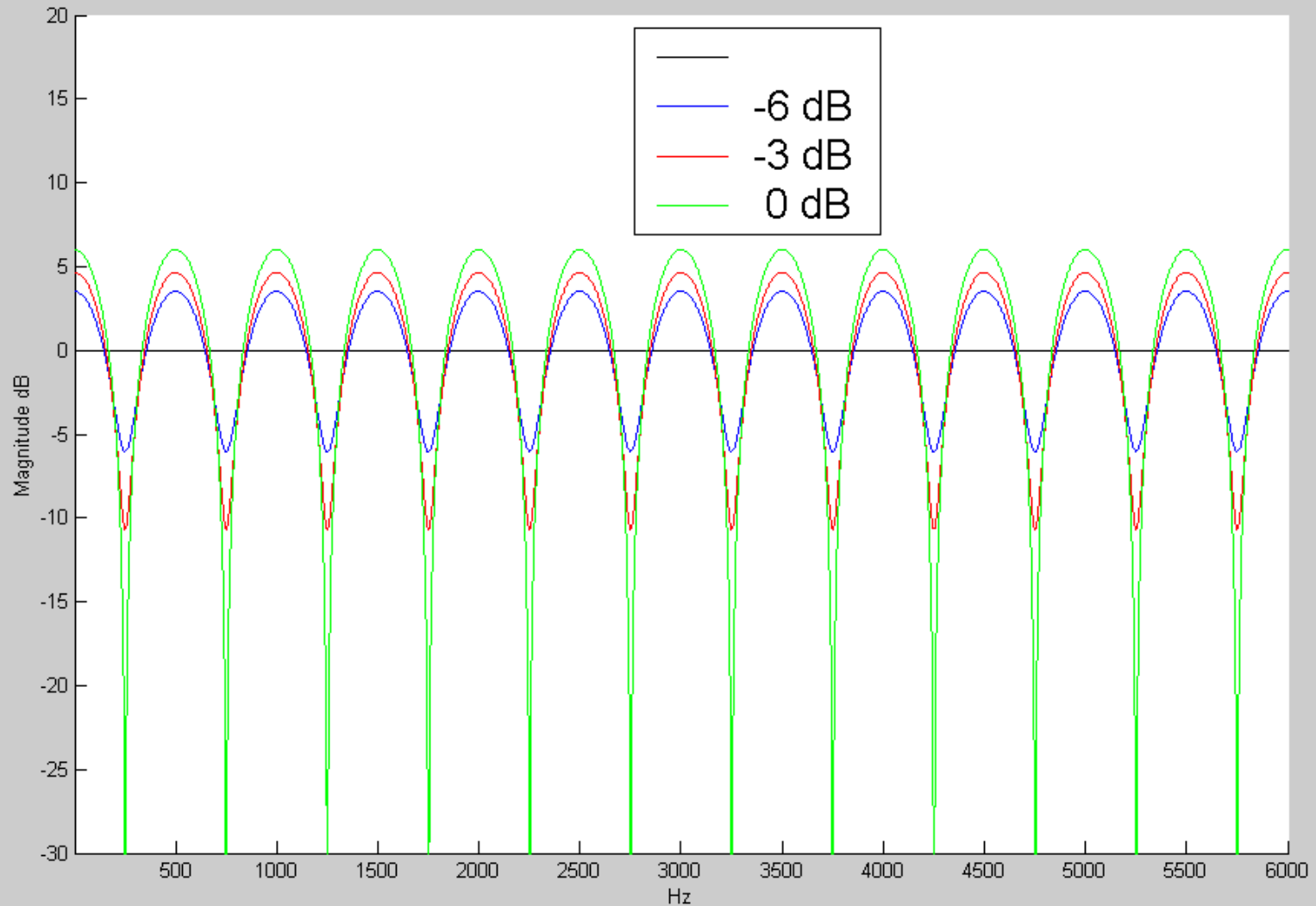
Simple case with one reflection that travels twice the distance as the direct path (no absorption, only inverse square law losses).

Open Loop Response

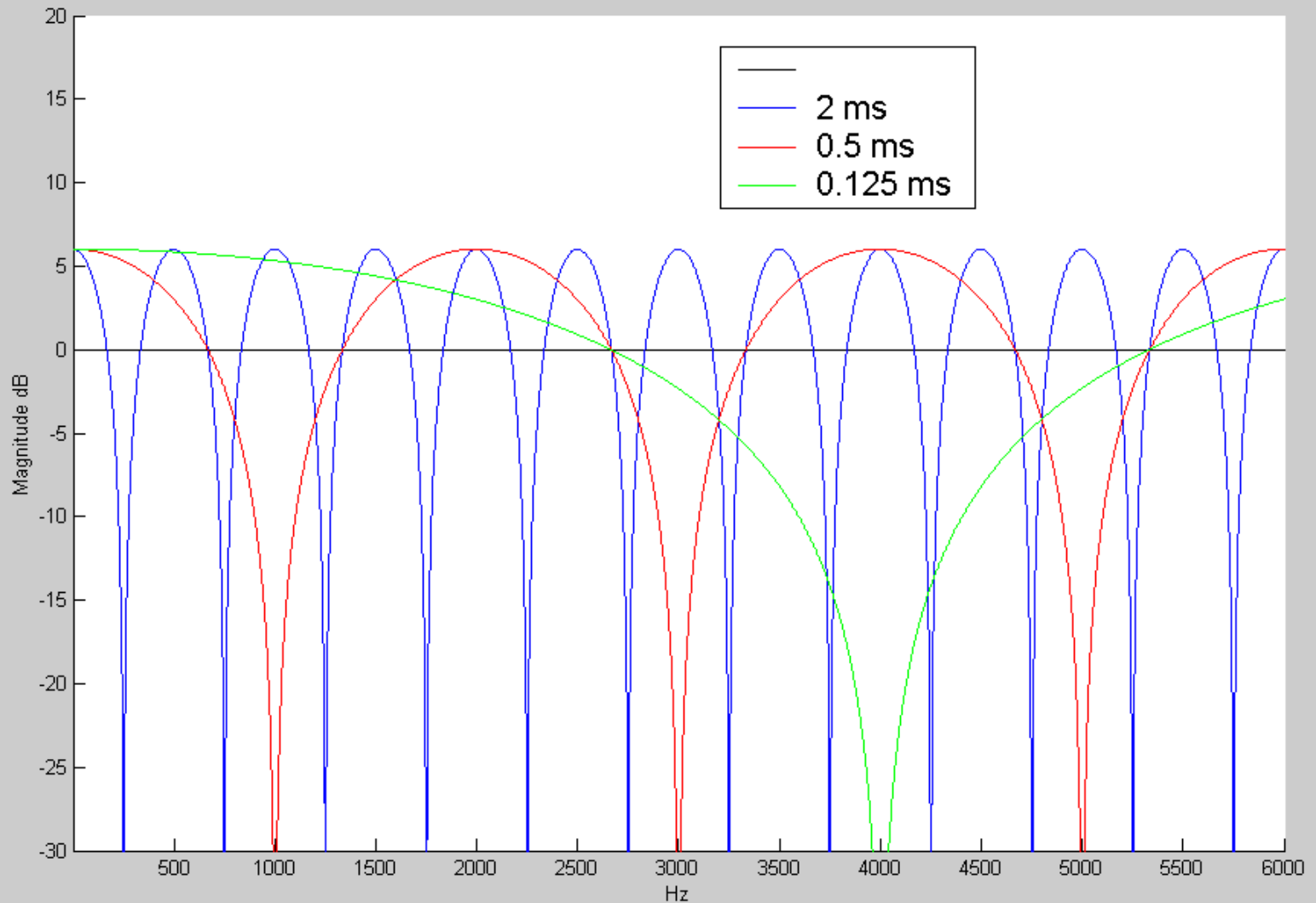
Comb filter effects due to a single reflective surface, delayed from the direct path by 2 ms. Amplitude of reflection is -6 dB down from the direct path.



Comb filter effects due to a single reflection or a 2nd source (speaker or microphone) , delayed from the direct path by 2 ms. Second source amplitudes of -6 dB, -3 dB & 0 dB down.



Comb filter effects due to a single reflection or a 2nd source (speaker or microphone) , delayed from the direct path by 2 ms, 0.5 ms and 0.125 ms. Second source amplitude is 0 dB down.



What happens when multiple reflections combine randomly with the modal response of the room?

“Zis is Kaos”



“Get Smart” 1965-1970



Feedback Characteristics

- Increasing system delay, increases the number and reduces the spacing, of potential feedback frequencies.
- System delay affects the rate at which feedback will grow and decay
- To bring a feedback frequency back into control, the open loop gain simply needs to be reduced below unity. However, dropping the gain just below unity can cause a very slow decay.
- As temperature increases, potential feedback frequencies shift higher. (the higher the frequency, the greater the shift)
- Multiple microphones, speakers or reflective surfaces can add unintended feedback as multiple acoustic paths interact. Temperature changes can shift these “hot magnitude spots”.

Now that we understand better what affects feedback

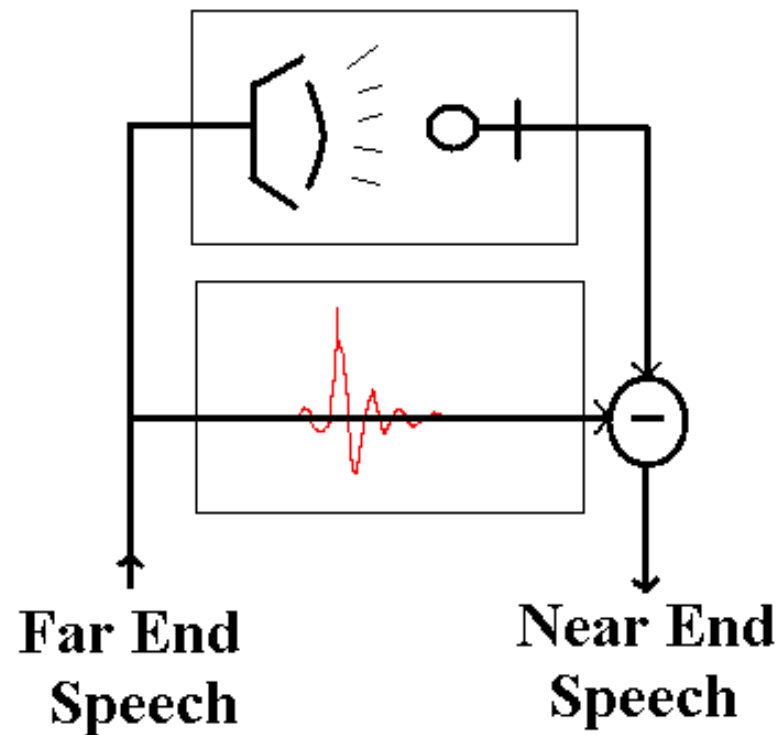


Maxwell smart secret agent 86
the best known member of **control**
www.wouldyoubelieve.com

... how do we control it?

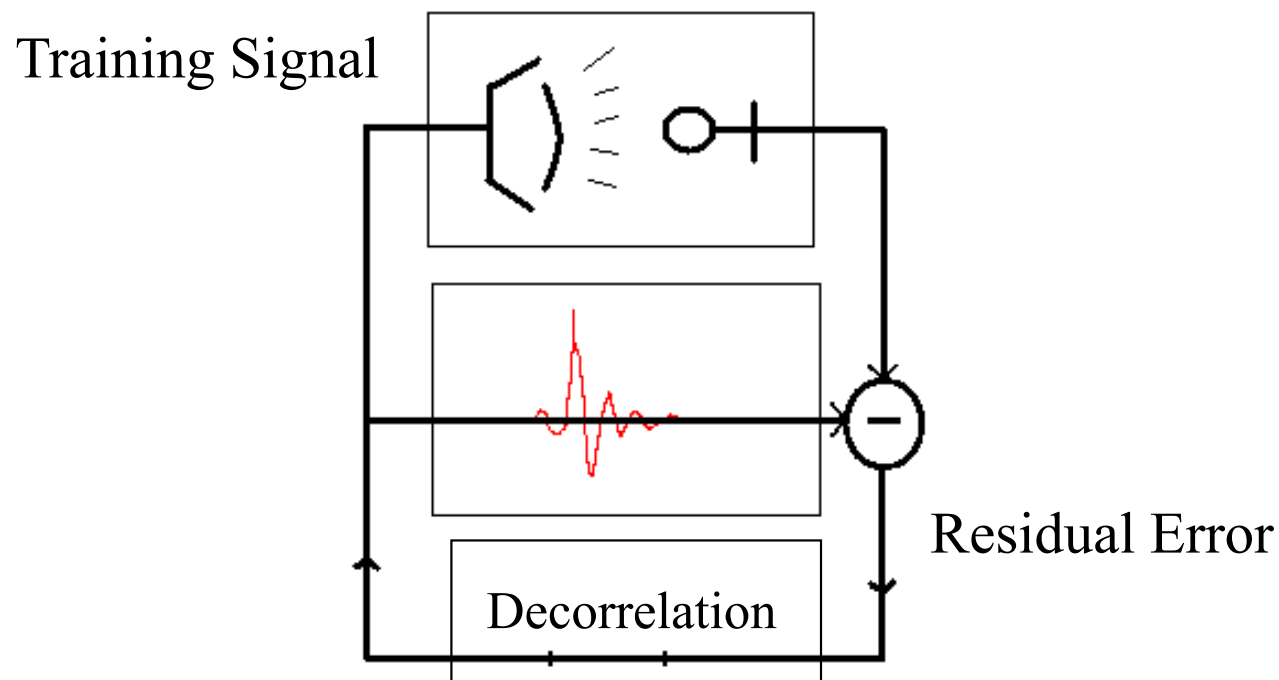
Method #1: Adaptive Filter Modeling

Widely used in teleconferencing applications to get rid of acoustic echo.



However, in acoustic echo cancellation the feedback loop is open, in sound reinforcement the feedback loop is closed.

This means that the residual error from the adaptive filter is highly correlated with the training signal. Adaptive filtering algorithms don't like that. So they will add a decorrelation function (such as a frequency shift) to help.





Method #1: Adaptive Filter Modeling Issues

- If the adaptive model is not perfectly accurate then distortion gets added. (This can include additional feedback when feedback would otherwise not be present)
- The needed decorrelation will also add distortion.
- Uses a considerable amount of processing power.
- In the real world this approach can give 10 dB or more of added gain before feedback.
- Most applicable for speech based systems.

Method #2 All pass Filters

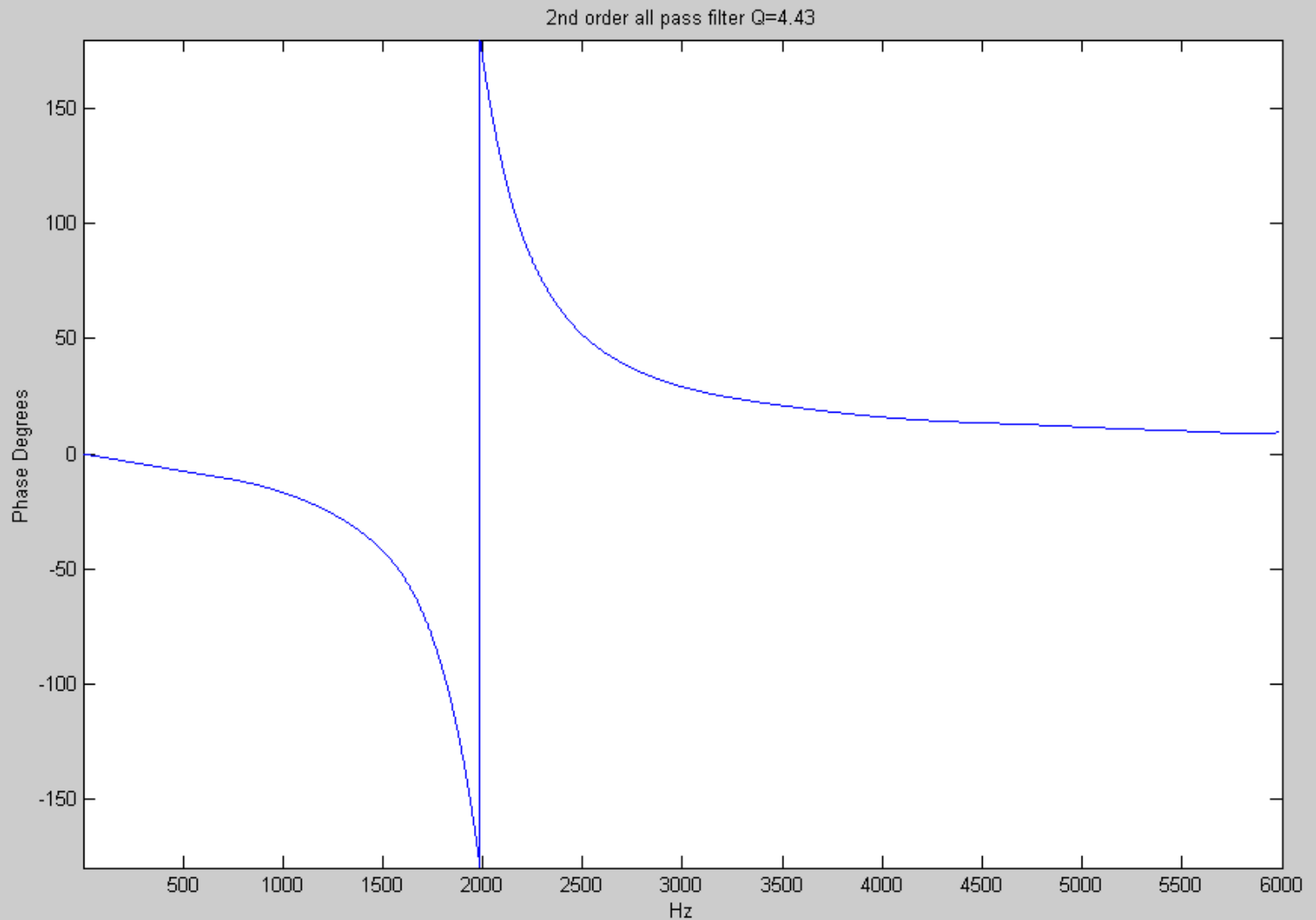
Can phase alone be used to control feedback?

A 2nd order all pass filter will introduce 180 degrees of phase shift at the center frequency, essentially turning a feedback resonance into a null. Which is exactly what we want.

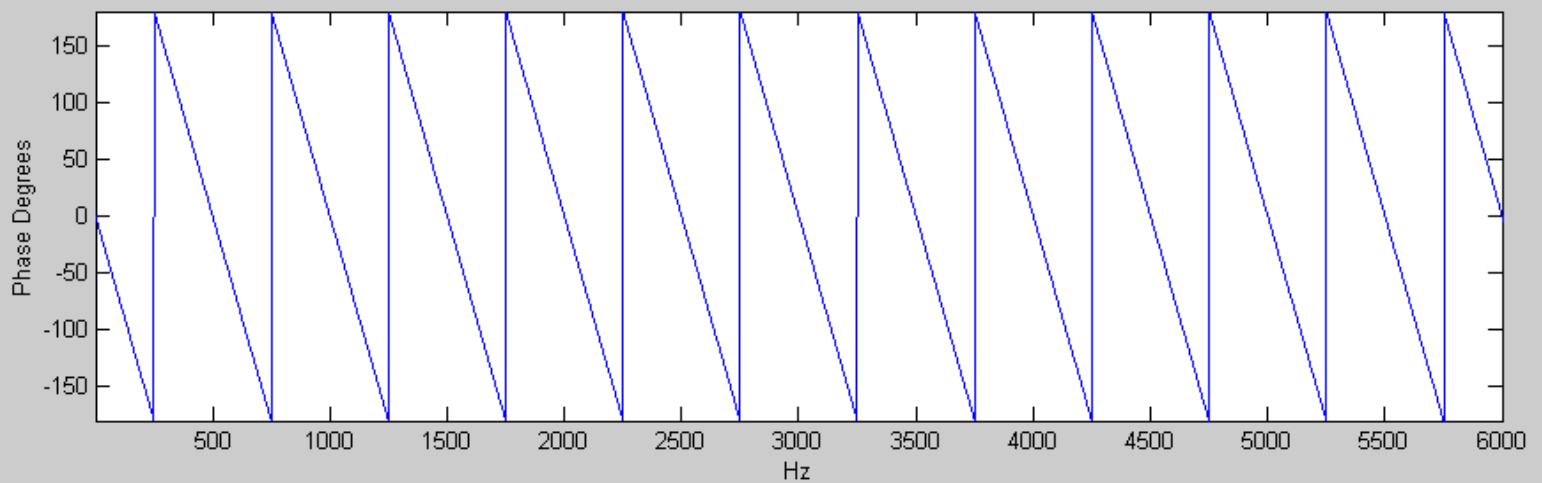
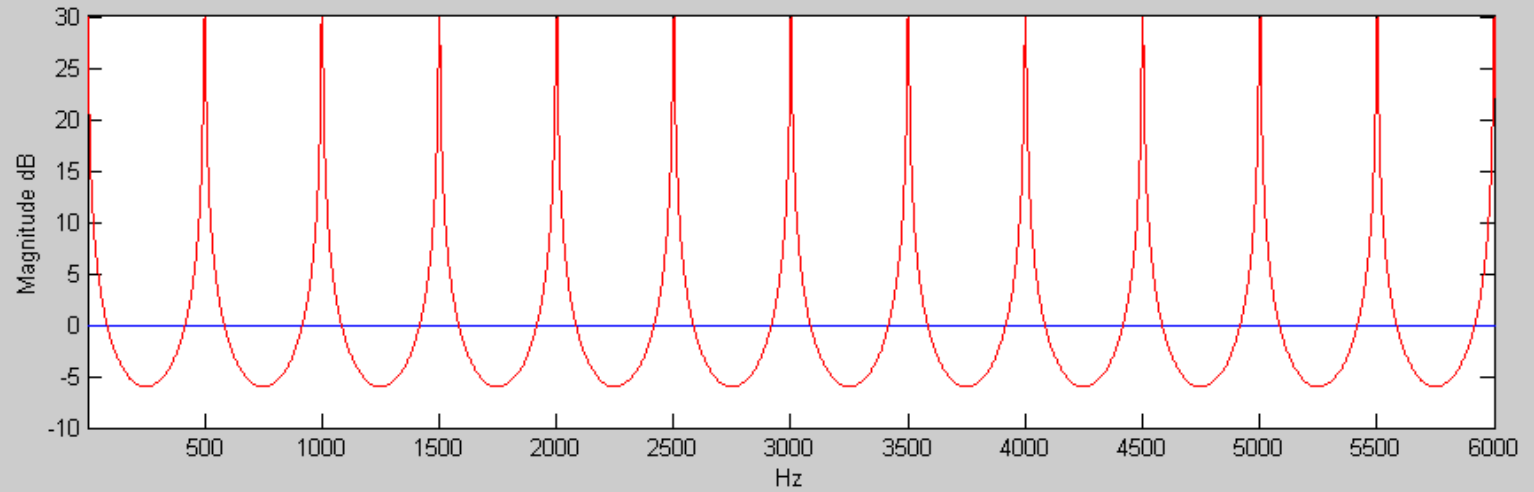
But ... but what else does it do?



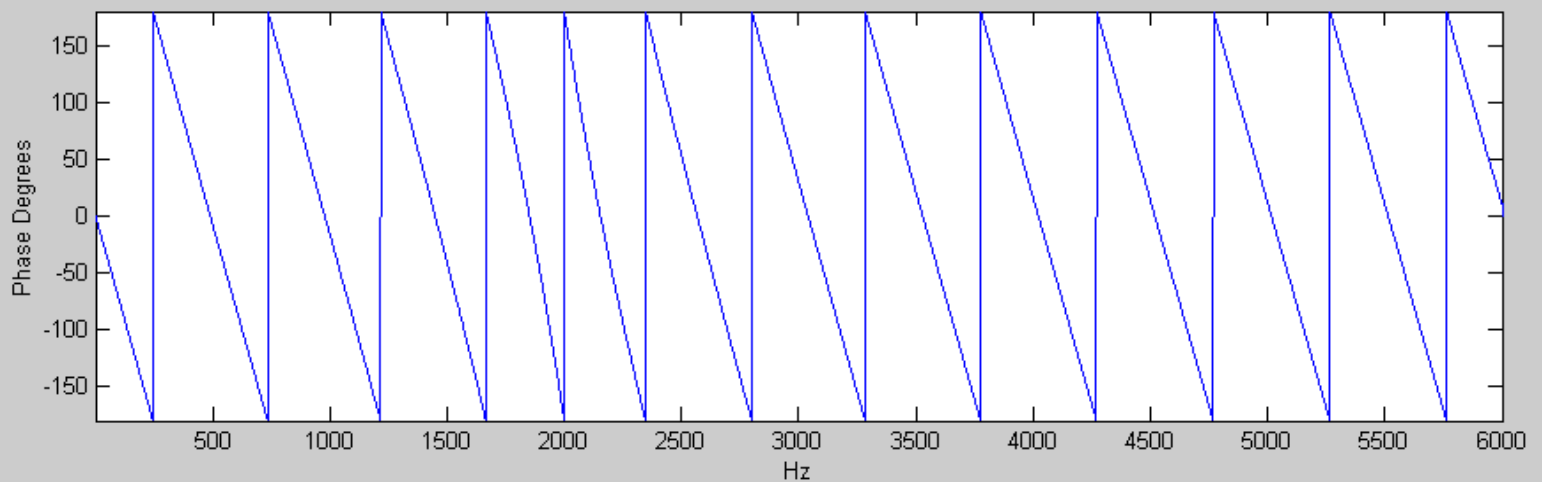
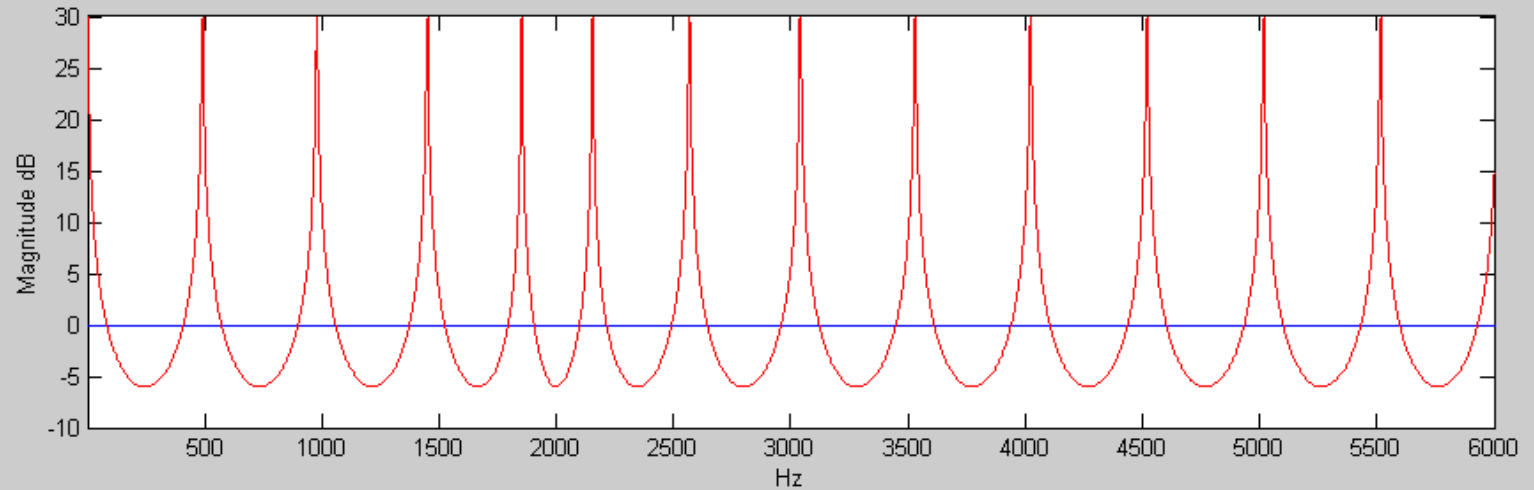
Along with the desired 180 degrees of phase at the center frequency there is additional phase introduced at adjacent frequencies.



Lets use our all pass filter to null the 2000 Hz feedback frequency



The 2000 Hz component is attenuated but 2 new potential components take its place.



If instead of a flat magnitude response, we had a peak at 2000Hz, then the all pass filter could 'chase' the potential feedback frequency to a neighboring frequency where the magnitude was below unity and bring back stability.

However, if the open loop magnitude is relatively flat then we have the Push down here ... and pop up there... Syndrome





Method #2: All pass filter issues

- Most applicable for a static installation with low delay or a resonant peak in the magnitude.
- Although it can fix a troublesome spot, it will change the potential for feedback at other frequencies.
- Uses minimal processing power.
- Its another tool in the toolbox however, it has limited utility.



Method #3 Frequency Shifting

What if you shifted the phase continuously, so as to decorrelate the input from the output?

$d\theta/dt$: is the definition of frequency

By constantly adding phase you create a frequency shift.

How much of a frequency shift is required to control feedback?

Actually the real question is how much can you get away with before people start to complain?

A 5-7 Hz shift is commonly used.



Method #3: Frequency shifting issues

- Completely eliminates the characteristic howl of feedback, however it trades it for a chirp.
- Provides a “couple” of dB of additional gain before feedback.
- Uses a moderate amount of processing power.
- Provides a really cool toy.

Method #4 The Auto Notch

Acoustic feedback is both a **magnitude** and **phase** problem.

Can we turn the gain down in selective areas and not lose overall volume and still keep things stable?

Analog notch filters have been used to control feedback since the 1970's.



The advent of digital signal processing enabled tighter feedback control.

Next we are going to compare and contrast the auto notching algorithms used by various feedback suppressors.



Two significant elements of autonotching:

Frequency detection

Fourier Transform (FFT and Others)

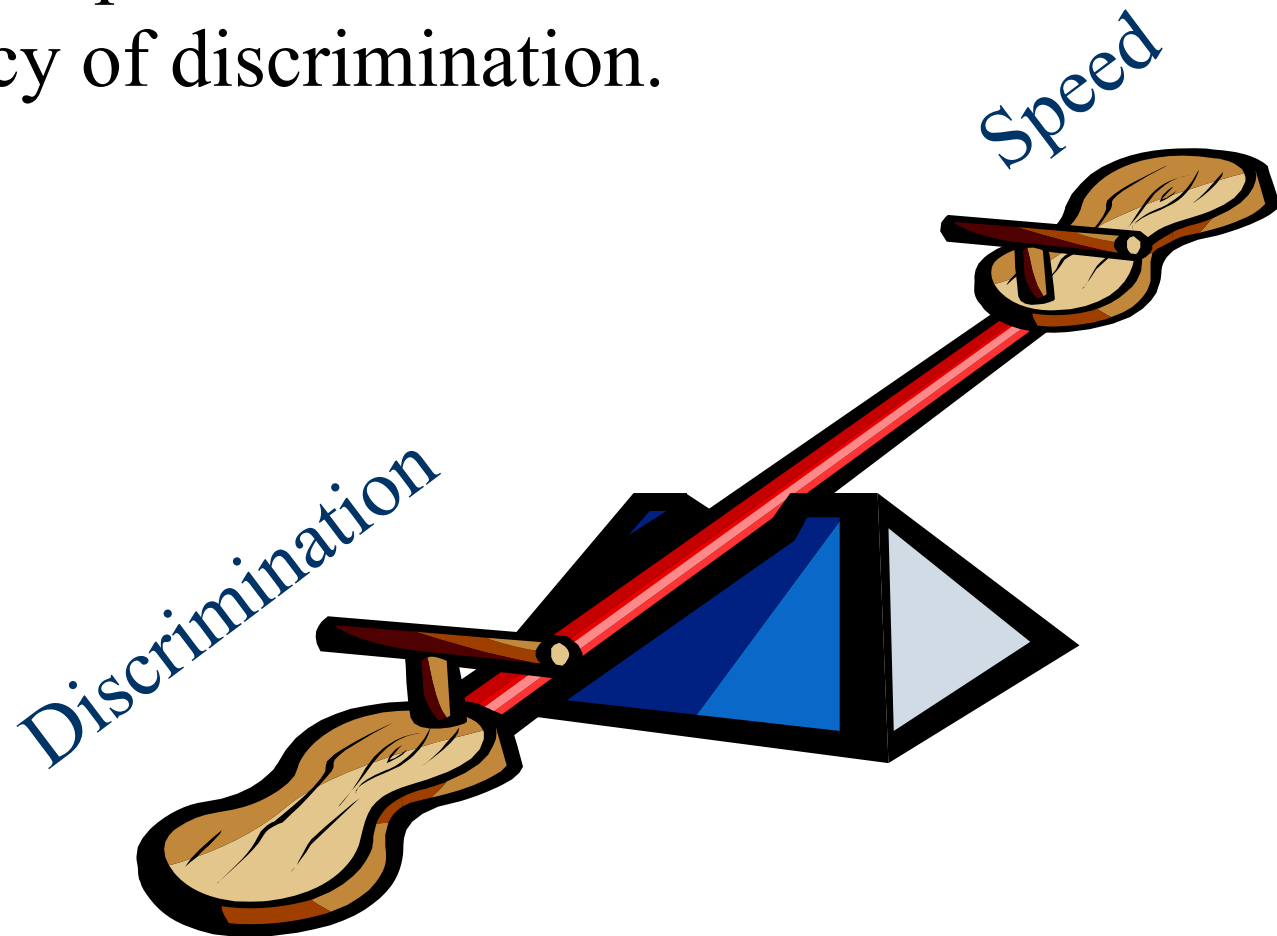
Adaptive Notch Filter

Feedback frequency discrimination

Music is rich in harmonics, so use presence of harmonics to discriminate feedback.

Use the characteristics of feedback such as consistent growth/decay rates to discriminate feedback.

Not surprisingly there is a trade off ...
between speed of detection and
accuracy of discrimination.



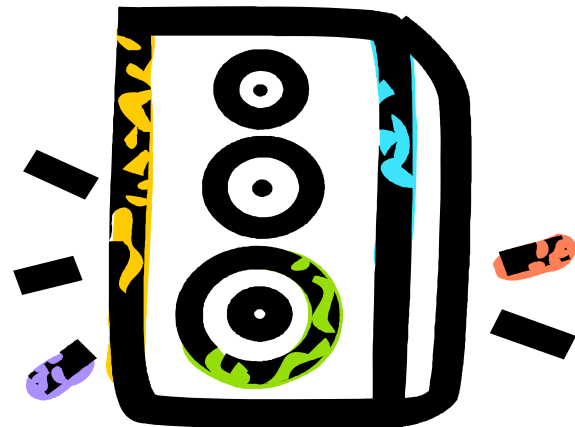
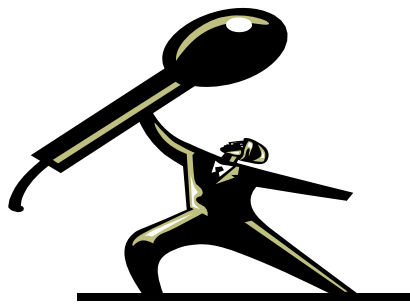
Some manufactures give user controls to adjust
the trade offs.

There are three difficulties in using the lack of harmonics to identify feedback.

#1

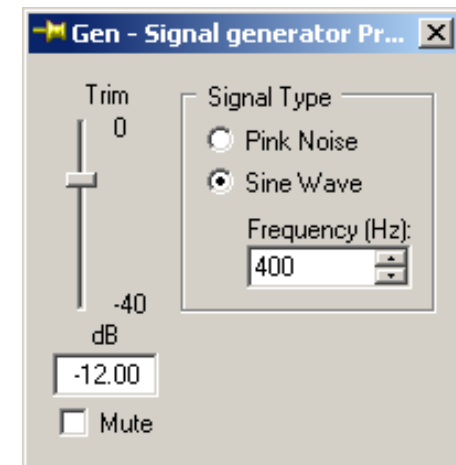
Feedback is perpetuated by transducers which can be rather non-linear especially if driven by a strong sinusoidal frequency. They will generate harmonics.

If a sinusoidal signal gets clipped harmonics will also be generated (even without the transducer).



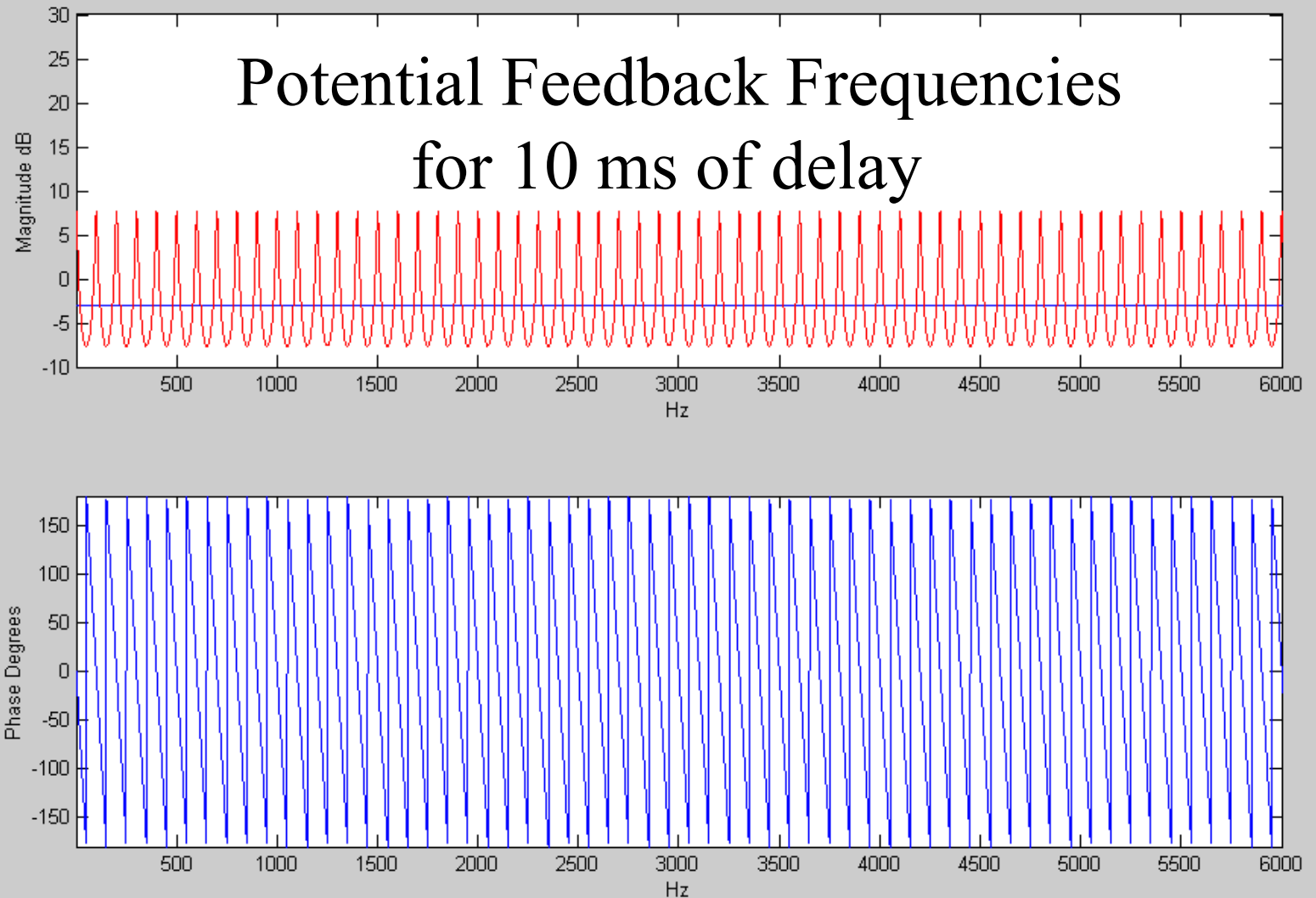


#2 Some non feedback sounds naturally have weak harmonics and will trigger a notch.



#3

It is not impossible to have the natural system transfer function cause 2 feedback frequencies that are harmonically related.



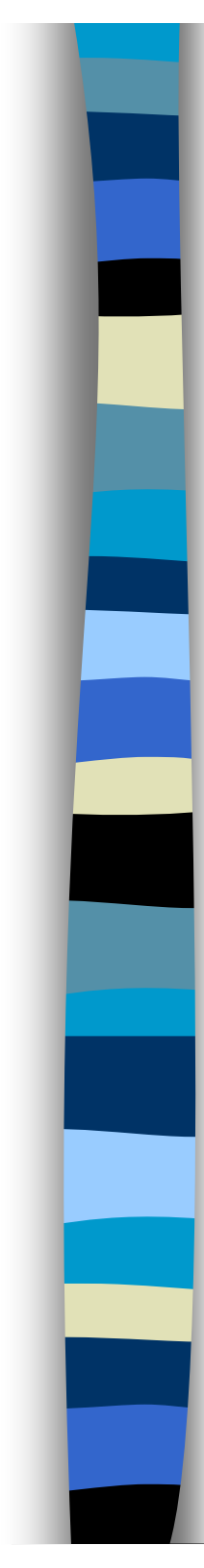


Are there difficulties in using the characteristics of feedback for discrimination?

Some of the dynamics in music can mimic the dynamics of feedback.

However, when the strength of a consistent frequency and its associated growth are combined with the decay coincident with the placement of a temporary notch accuracy improves .

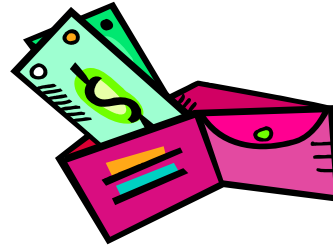
Overall it is a more workable solution then relying on harmonic content.



All of the manufacturers give a fair bit of flexibility in deciding notch quantity, bandwidth and depth.

They also give the ability to select whether or not a filter will remain fixed (static) after the initial placement or if it will be floating (dynamic) so as to allow filters to be recycled.

Here are some final tips



- ❑ Place notches last (after all system EQ and delay changes).
- ❑ Don't use one box to auto place notches and then copy settings to another box that has different delay or processing in it.
- ❑ Avoid deep narrow notches, use shallow wide instead (remember you only need to bring the open loop gain below unity gain to keep things stable).
- ❑ Remember the temperature shift effect. Higher frequencies are more susceptible to phase differences caused by temperature and delay changes.
- ❑ Only keep static filters that you have convinced yourself are necessary.
- ❑ Only leave a couple of floating filters active and restrict their depth.