

This project was motivated by the fun of digging CW weak signals out of the noise. Copy of weak signals is a skill necessary for Low Bands (160m), VHF, microwaves and EME CW operators. Yes, I am 30 years late with the project, but better late than never. It is fun and a big motivation for our hobby.

3S proved to be very efficient resulting in super narrow CW filters 5 Hz to 10 Hz BW, with Q between 40 and 50, and no ringing for high CW speeds.

The pursuit of narrow filters to improve signal to noise ratio, end up with new digital modes with effective filter bandwidth (BW) near or below 1 Hz. Narrow BW proved itself to be very efficient, much better than the human brain. But if you have a brain and enjoy the task and the fun digging weak CW signals out of the noise, this project is a delight and enjoyable way to experiment with CW filters with BW below 10 Hz.

The human ear is very selective to audio frequencies, keeping the receiver BW large, like SSB 2.4 KHz, your ear works like a narrow CW filter too. The ear filter has a minimum BW of 90 Hz near 200 Hz, it is called critical frequency, below that frequency there is no improvement, and above it, the BW is wider, like 120Hz at 1 KHz, for CW it is not a big difference. To notice a signal to noise ratio (SNR) improvement you really need to use filters below 100 Hz.

Here is the complication, the narrow filter needs to have sharp edges, that's bring two things, ringing and intermodulation noise or commonly known as phase noise of the filter edges, (frequency group delay). Not getting into the math involved, it is bad because the problems increase as you increase the CW speed, creating a CW speed limit.

The first successful implementation of 10 Hz BW filter was introduced with coherence CW in the 90's, the speed was limited to 12 wpm, PSK31, and other digital modes were a development of this process.

The Audio Peak Filters on the new SDR radios are a great improvement, with practical $Q = 30$. However, it has the same limitation of ringing, and increase of noise, also limiting the usable CW speed.

The 3S is a mechanical, acoustic device that does not ring. How is that! Well, it is not really a filter, it works like a filter, but it is indeed an oscillator. The speaker creates a disturbance into a column of air inside a pipe that is resonant based on the length of the pipe, it works like an organ pipe, or a flute. The phase of the signal is important to keep the pipe singing. The RF filter BW needs to be kept between 400 and 200 Hz, to avoid the problems mentioned above.

The 3S practical results are amazing and delightful to use. The way I use it is to implement a concept developed by NASA in the 90's too. It is called 3-D sound. The idea is to present the sound coming from two different sources 45 to 60 degrees from your head. I installed the two 3S with the same center frequency behind my PC monitors, they can stand vertically or horizontal, that way the sound is presented in a natural manner. The ears and the brain, the human audio system, working together to process the signals. Using two sources equally distanced the sound seems to be coming in front of you, or changing the phase of one source moves the position of the sound, like a surround sound. This process allows the listener to selectively focus on one sound coming from a specific direction. It is amazing, like you can disregard the noise of the linear amplifier coming from a different direction, better than a loved headset. Like at a party or restaurant, you can focus on one conversation out of many. 3-D sound also improves intelligibility.

The new speaker drivers used on this project is an evolution of NASA 3-D implemented on TV sound bar that simulates a 5-1 system, changing the phase of the sounds.

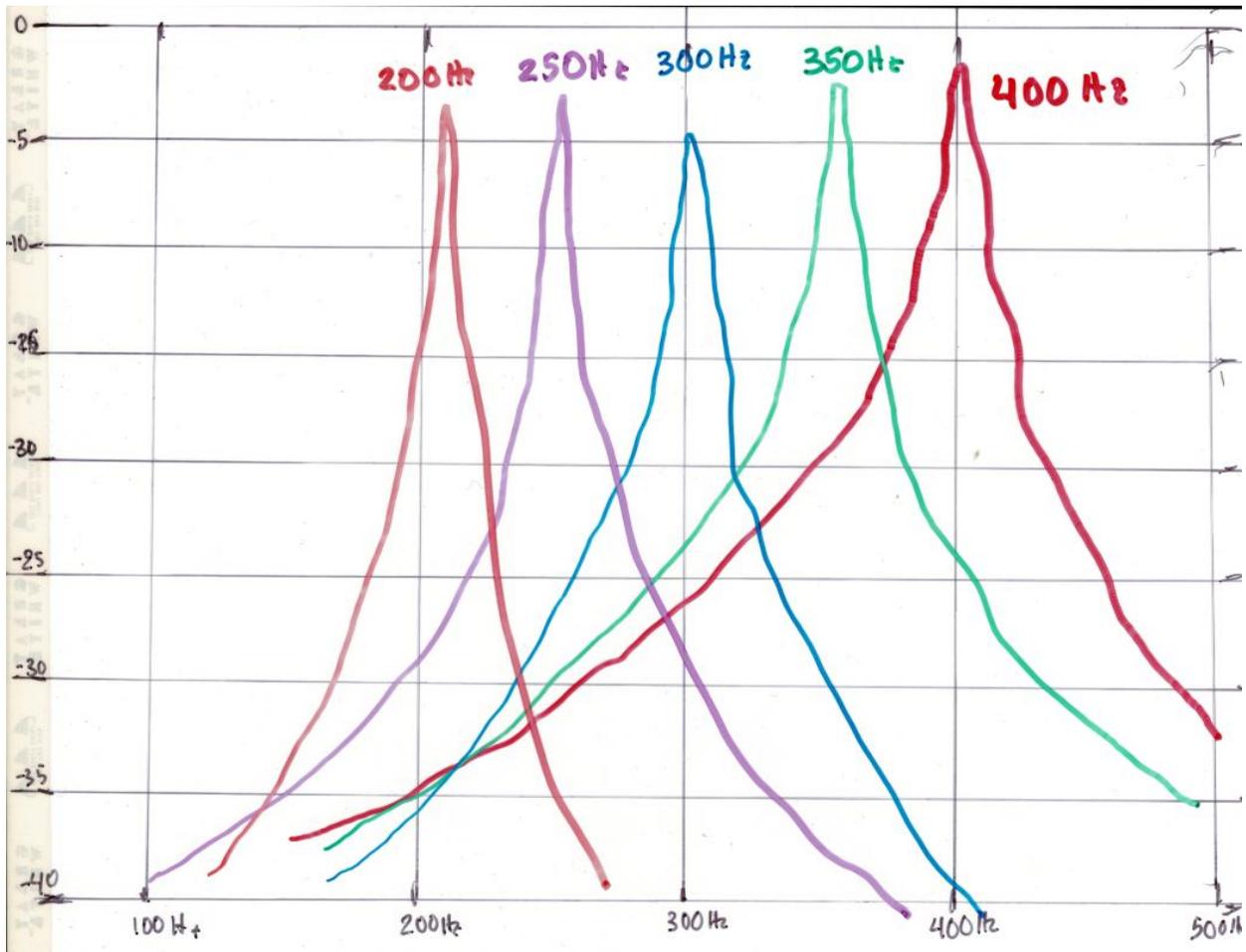
Few new drivers have a flat response inside a small box, most of them present a large resonant peak response that needs to be matched with the filter center frequency. Both speakers are usable, but the flat response gives the best Q, like the PeerLess TC7FD. But most speakers like to work between 250Hz and 400 Hz.

The logarithm center of the human ear is 630 Hz and the best frequency for CW accuracy is near 500 Hz. For most cases where the source of noise is not atmospheric, 600 Hz is probably the best choice for CW copy. When atmospheric noise is present, like 160m summer season, there is another complication named "masking". In short, a louder low frequency sound lowers the perception intensity of a higher frequency sound. If you once operated 160m during summer, you are aware of the experience. Each crash knocks you down for several seconds. The only way to reduce masking is to lower the CW pitch, making the QRN soft, my great friend and Elmer PY1RO used to hear CW at 200 Hz pitch. Rolf earned the first DXCC (#14) outside the USA and south of the equator, it means DX season in the summertime. Bill W4ZV mentioned he prefers 270Hz and likes 240Hz. Most low band operators like 350 Hz cw pitch.

CW pitch to improve SNR is not about how the CW notes sound, the main thing is how the noise sounds. Like most of the active APF filters react to the QRN with a lot of intermodulation noise at the same pitch of the CW signal, and rings. The 3S system improves the audio level SPL (sound pleasure level) by + 10 db, and the attenuation outside the bandwidth by -30 db or more. Depending on the center frequency.

Lower the cw pitch also helps with BW because the Q is the same, so 200Hz divided by Q = 50 is 4 Hz BW, 600 Hz with Q = 50 BW is 12 Hz. Here are the summary results of my 3S system

N4IS 3S - APS		CW PITCH CENTER FREQUENCY Hz					
Feb 23, 2025		200	250	300	350	400	600
SPL	Q	43.5	41.6	44	50	50	24
+10	BW db	BW Hz	BW Hz	BW Hz	BW Hz	BW Hz	BW Hz
+7	-3	4.6	6	6.8	7	8	25
+4	-6	6.6	10	9.5	13	15	35
0	-10	10.7	16	20	22	29	60
-10	-20	31	47	61	68	88	165
-20	-30	72	133	136	226	200	350
-30	-40	153	270	290	-	-	-
-40	-50	200	-	-	-	-	-

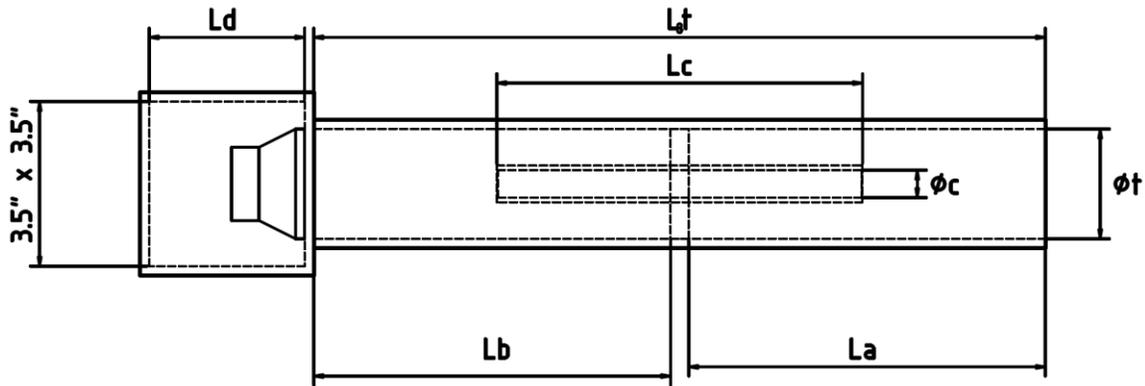


3S

S_{Super} $S_{\text{Selective}}$ S_{Speaker}

N4IS

$$L_a = L_b = L_c$$



Dimensions

L_t = Main tube total length

L_a = Main tube open side, inside length

L_b = Main tube speaker side, inside length

L_c = Inner tube length

L_d = Wood sealed box deep length (internal volume 3.5" x 3.5" x L_d)

\varnothing_t = Main tube internal diameter ~ 2" (50mm)

\varnothing_c = Inner tube internal diameter ~ 3/4" (20mm)

Resonances

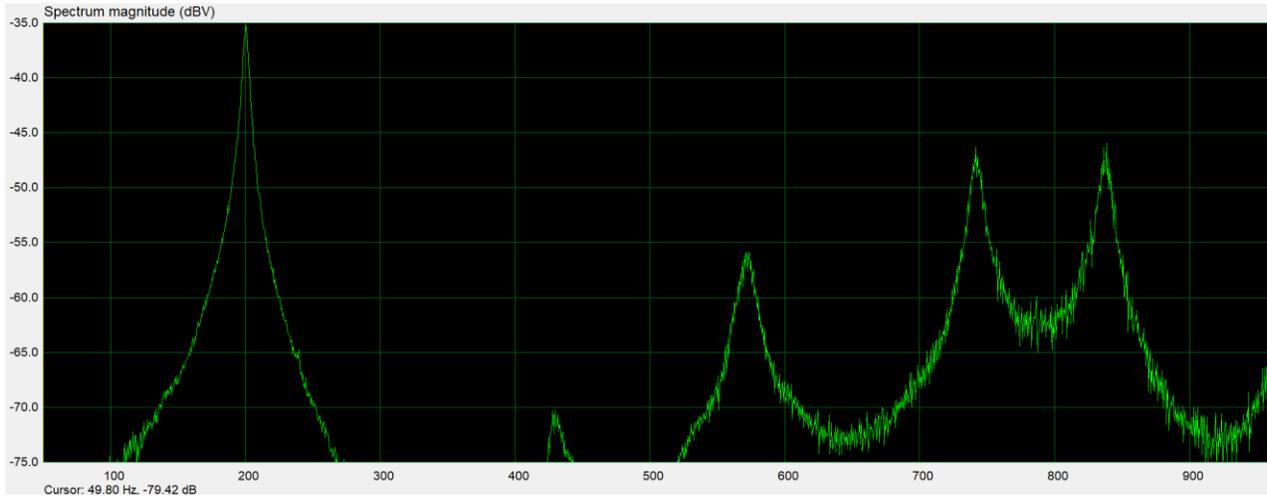
F_a = APS resonant frequency peak

F_n = Inner tube resonant frequency notch (2 x F_a , notch 2nd harmonic)

F_b = Speaker resonant frequency inside the sealed box

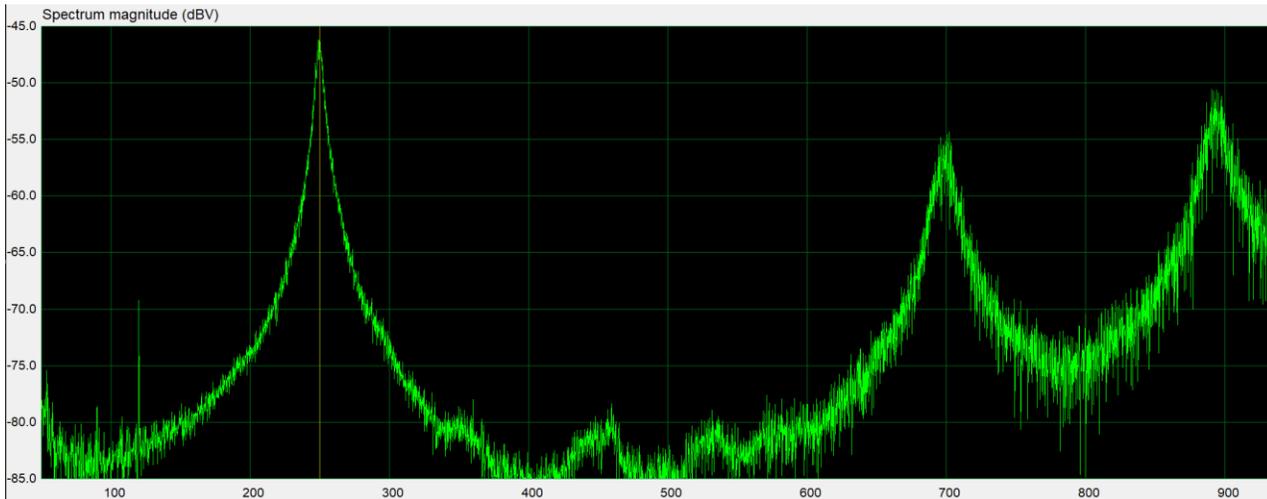
3S Fa = 200 Hz audio peak filter Lt = 914mm

Vert. 5 db Horiz. 100 Hz



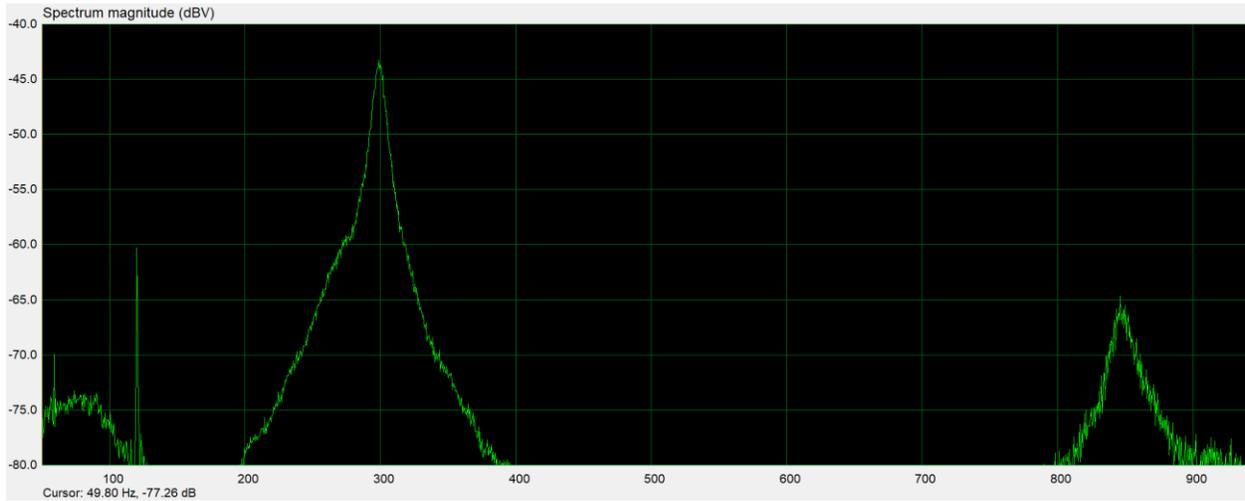
3S Fa = 250 Hz audio peak filter Lt = 726mm

Vert. 5 db Horiz. 100 Hz



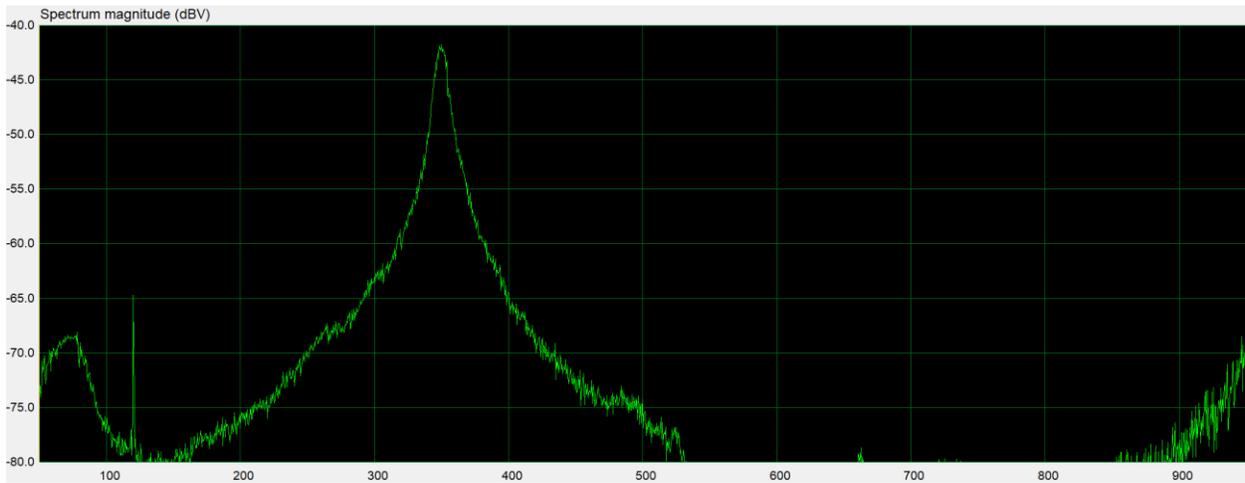
3S Fa = 300 Hz audio peak filter Lt = 602 mm

Vert. 5 db Horiz. 100 Hz



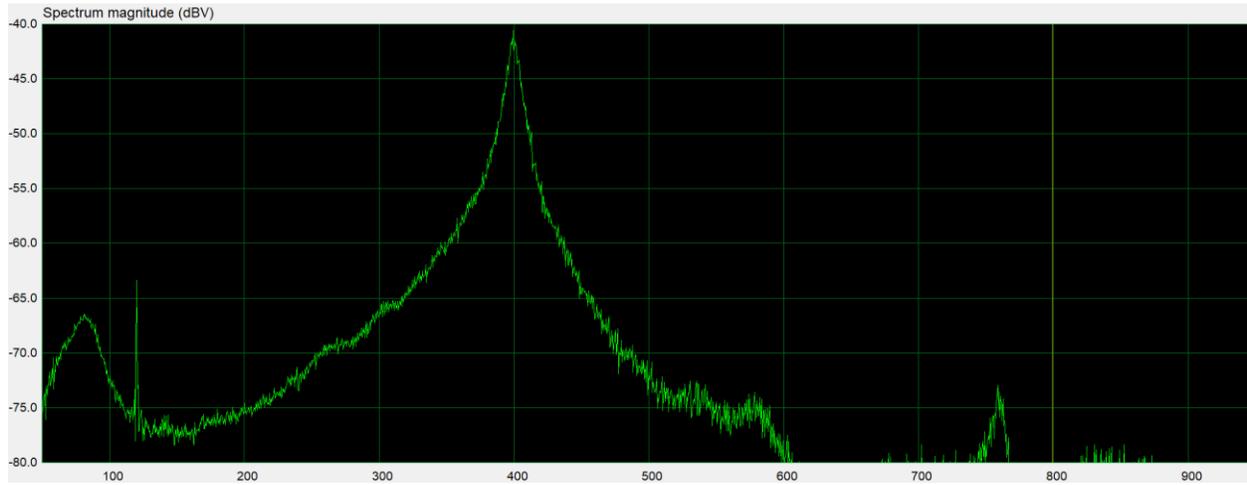
3S Fa = 350 Hz audio peak filter Lt = 520 mm

Vert. 5 db Horiz. 100 Hz



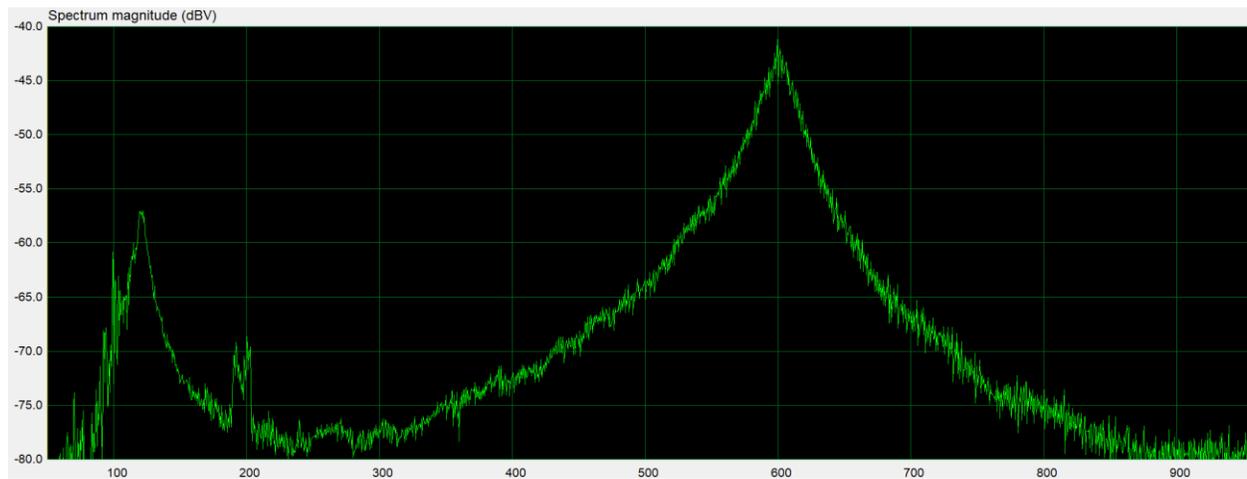
3S $F_a = 400$ Hz audio peak filter $L_t = 459$ mm

Vert. 5 db Horiz. 100 Hz



3S $F_a = 600$ Hz audio peak filter $L_t = 287$ mm

Vert. 5 db Horiz. 100 Hz



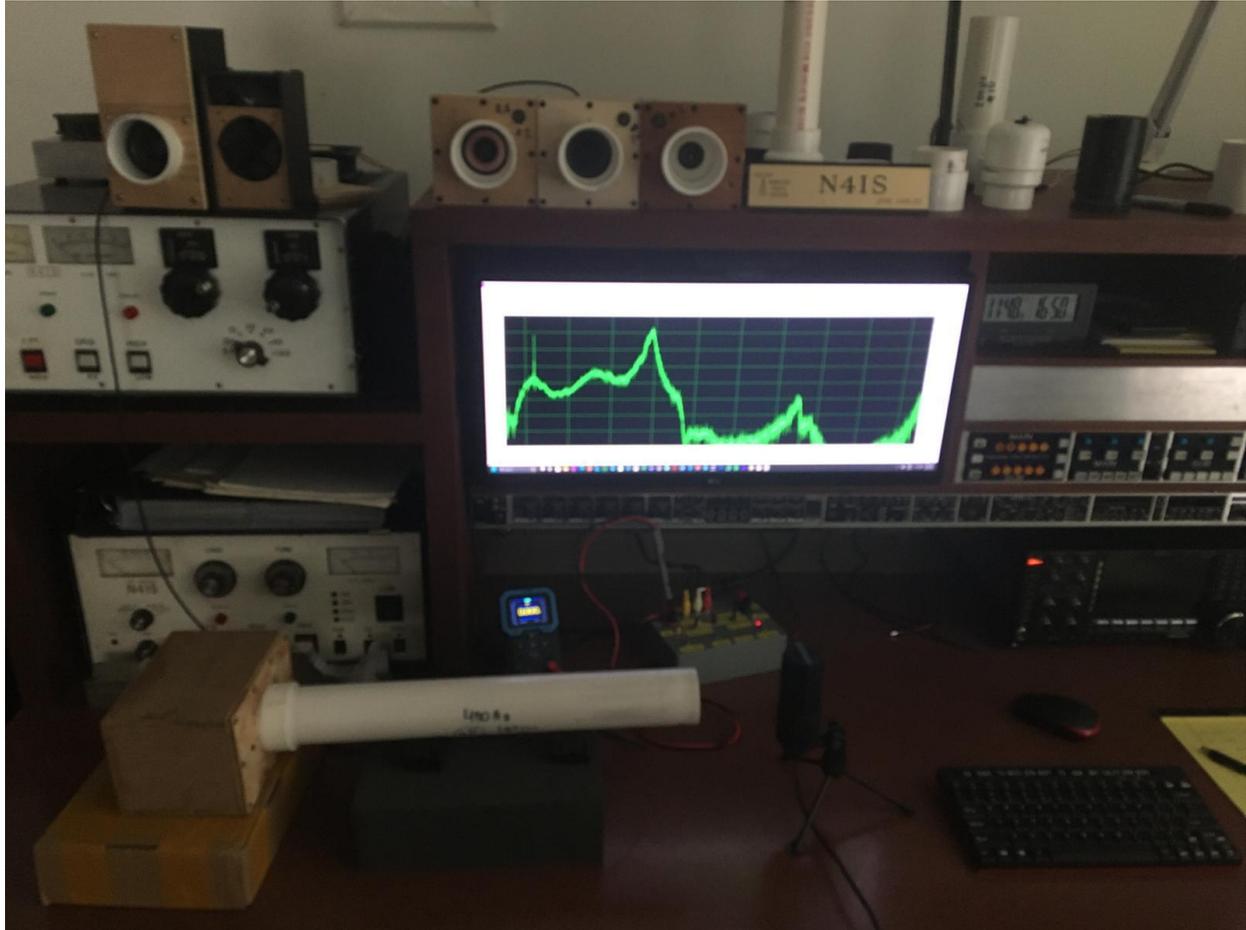
Biult and tested over 100 filters



Tested over 40 different speakers



A lot of bad results.



- *Speaker resonance inside the box was different from the filter center frequency.*
- *Speaker box leaking air.*
- *Speaker not flush with baffle, leaking air into the tube.*
- *The ring holding the inner tube leaking air.*
- *The best results with \emptyset near 2" and inner tube \emptyset near $\frac{3}{4}$ ".*
- *$L_a = L_b = L_c$, for maximum SPL output, and best bandwidth.*
- *Adjust the speaker box internal volume to match the resonant frequency, $L_d = 1.5$ ".*
- *All the above to get $Q=50$, +10 db SPL gain, and > -30 db attenuation.*

Finally, some very good results! $Q = 50$ (Fc/BW)

