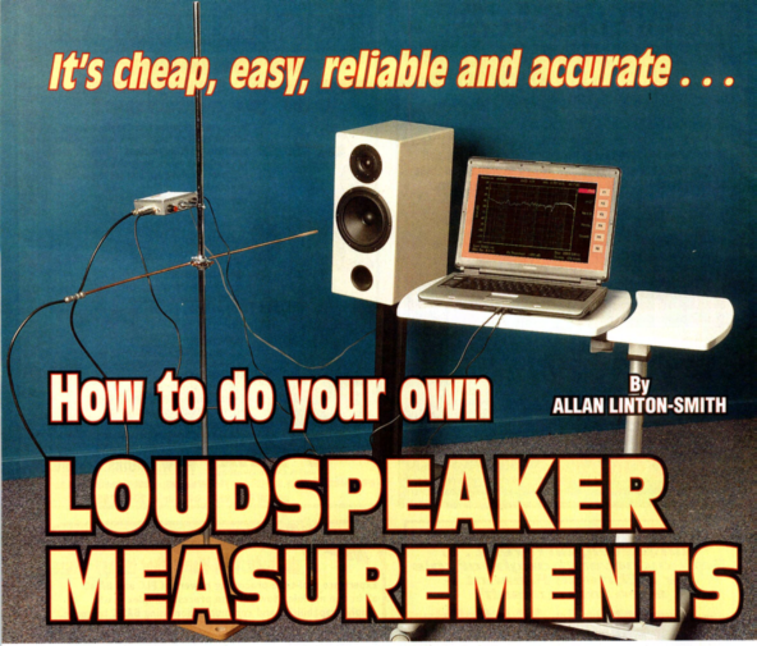


## It's cheap, easy, reliable and accurate . . .

# How to do your own LOUDSPEAKER MEASUREMENTS

By ALLAN LINTON-SMITH

***Measuring loudspeakers used to require a lot of expensive equipment, an anechoic chamber and a lot of skill. Nowadays we can do it easily with some low-cost software for the PC, an equally low-cost amplifier to drive the loudspeaker and a calibrated microphone which you can buy cheaply (or you can even make a your own to save even more money).***

**Accurate, commercial speaker measurement systems** can cost tens of thousands of dollars – way outside the budget of even the most dedicated audio enthusiast.

Now, with the advent of well developed PC "virtual instruments" and much-improved electret microphones, we are able to present an economic speaker measurement system capable of accurate and reliable results.

We have often seen enthusiastic loudspeaker experimenters take great care in selecting speaker drivers and mounting them in well-designed cabinets, only to find that the results don't live up to their listening expectations.

More often than not, they can be let down by incorrectly designed crossover systems which cause large peaks (or worse still, deep troughs) or incorrect level adjustments for tweeters and midrange drivers.

This project removes the subjective errors which may result from adjustments made by using only listening tests. The operator will also have a facility to print all response curves.

### The test set-up

An audio sweep signal from 20Hz to 20kHz from the virtual instrument is amplified and fed through the speaker under test (SUT).

A wide-range electret microphone set very close to the speaker picks up the swept signal and its output is amplified and fed to a "virtual" spectrum analyser which then plots the amplitude of the speaker response on the vertical (Y) scale versus frequency on the horizontal (X) scale using a principal known as Fast Fourier Transform (FFT).

The result is a plot of the frequency response of the SUT. In this case we are using a "virtual spectrum analyser" which you can purchase and download from [www.fatpigdog.com](www.fatpigdog.com)

The Author describes his Audio Spectrum Analyzer as suitable for "the Acoustic Specialist, Vibration Analyst, RF Engineer or True Geek"!

Even if you're none of those, you'll find the Audio Spectrum Analyzer easy to use and a very worthwhile program to own.

Best of all, at just $US39.99 the software is very reasonably priced but with the volatile Aussie dollar at the moment we won't even hazard a guess at the $AU price; we imagine it will be fairly close to the $US price.

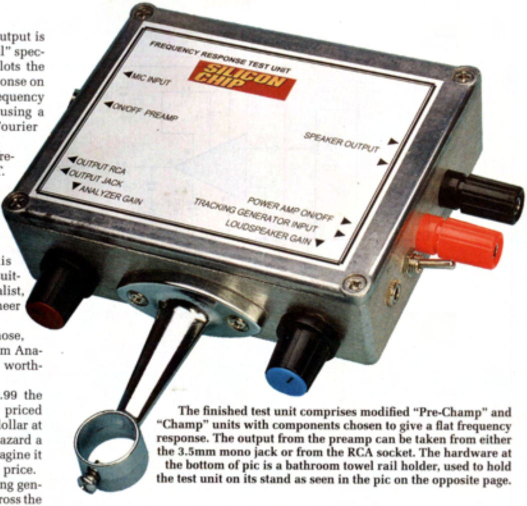
It also has a built in "tracking generator" (TG), which sweeps across the desired frequency range, in step with the analyser.

The audio sweep signal is fed to a "Champp" amplifier (SILICON CHIP, February 1994). This "oldie but a goodie" has been modified to give a flat frequency response and can drive an 8-ohm speaker to about half a watt.

This may not seem very much but you will be surprised how loud it can be and it is certainly adequate for frequency response testing. Of course, you could use any power amplifier which has as good or better response than the modified "Champ" which is ±0.2dB from 20Hz to 20kHz

The signal from the loudspeaker under test is picked up by a specially built microphone or a commercial calibrated microphone. We'll have more details on these later in this article.

The electret then feeds our "Pre-Champ" preamplifier (SILICON CHIP, July 1994) which has also been modified for a flat response. The resultant signal is fed to the spectrum analyser for processing. You can save and print your response curves for further analysis. Both the "Pre-champ" and "Champ" are mounted in the same diecast box but each has a separate battery




**The finished test unit comprises modified "Pre-Champ" and "Champ" units with components chosen to give a flat frequency response. The output from the preamp can be taken from either the 3.5mm mono jack or from the RCA socket. The hardware at the bottom of pic is a bathroom towel rail holder, used to hold the test unit on its stand as seen in the pic on the opposite page.**

## Specifications:

| | |
|---|---|
| Microphone frequency response: | (31.5Hz-20kHz) ±2dB |
| | (31.5Hz-16kHz) ±1dB |
| | (20Hz-20kHz) ±2.5dB |
| Preamplifier frequency response: | ±0.2dB (20Hz-20kHz) |
| Power amplifier frequency response: | ±0.2dB (20Hz-20kHz) |
| Power amplifier output (before clipping): | 200mW into 8 ohms |
| Frequency response of virtual instrument: | ±0.4dB (20Hz-20kHz) |
| Overall measuring accuracy: (*without calibration chart*) | ±2.9dB (20Hz-20kHz) |
| Overall measuring accuracy: (*using calibration table*) | ±1dB (20Hz-20kHz) |
| THD+N preamplifier: | 0.1% at 1kHz (22Hz-22kHz). |
| THD+N power amplifier: | 0.4% at 1kHz (22Hz-22kHz) 250mW |
| Crosstalk from pre-amp: | -63dB at 1kHz, 20mV input |
| Crosstalk from poweramp: | -47dB at 1kHz, 20mv input |
| Preamp input maximum: | 50mV |
| Preamp input minimum: | 10mV |
| Power amp input maximum: | 500mV |
| Power amp input minimum: | 30mV |
| Preamp phase distortion: | ±6.35° (below 200Hz). |
| Preamp intermodulation distortion: | 0.1% (88mV output 70Hz/7kHz). |
| Preamp signal-to-noise ratio: | -107dBV (10Hz-80kHz ref 630Hz 25mV) |
| THD+N tracking generator: | 0.0066% at 1kHz (22Hz-22kHz) |

(*using Acer Aspire One model KAV10 with Windows XP*)

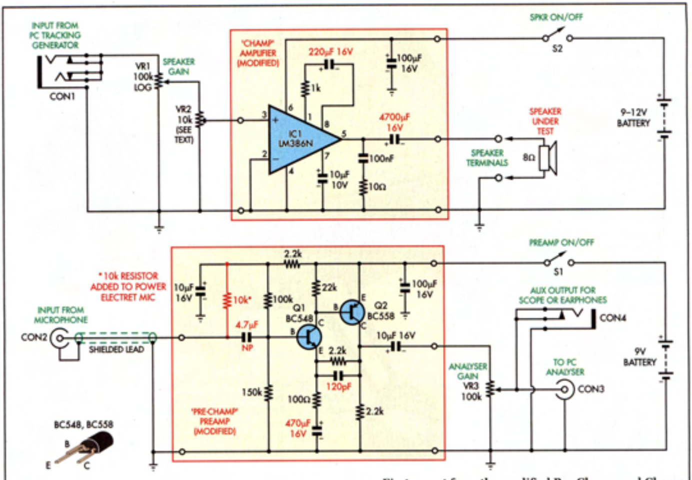



**Fig.1: apart from the modified Pre-Champ and Champ projects, the rest of the Analyser is simply input and output connections. The software that drives it all – fatpigdog – is powerful but quite cheap.**

to minimise crosstalk and feedback.

Not only is the setup useful for measuring loudspeaker frequency response, it can also be used to plot the frequency response of an amplifier, pre-amplifier, audio filter or crossover network.

It is also handy as a general purpose portable microphone for public address systems or DJ work or even for good quality recording – just plug it into any line input or power amplifier.

Also, if you plug it into a frequency counter, you will be able to accurately tune instruments (assuming you know or find out what frequency equates to the notes in your particular instrument).

The virtual spectrum analyser will also be very useful as a training tool because it has been specifically designed to look and feel like a typical bench top analyser.

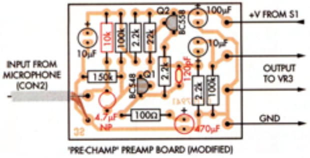
The new tracking audio generator included in the fatpigdog software is very useful too. It measured 0.0066% THD+N (at 1kHz when set at 635mV on "zero span"; measured on an Audio Precision test set!).

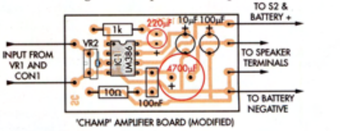
The THD+N is largely up to the quality of your sound card although ours was measured from the standard sound card in an Acer Aspire One which cost less than $500.

You can also use it all as a spectrum analyser and waterfall analyser and play around with various colour modes. It requires some skill and patience (just like a real benchtop spectrum analyser) but if you experiment, you will learn to master it all fairly quickly.

## Construction

Assuming you're building the Pre-champ and Champ from kits, start




**Figs.2&3: Pre-champ and Champ PCB component overlays with the changed components (from the original projects) shown in red.**

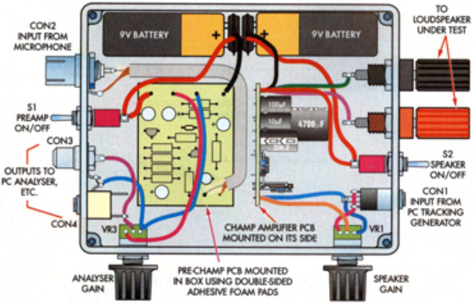



**Fig.4: use this assembly diagram in conjunction with the photo below when you put it all together. The two PCBs are secured to the case with double-sided foam adhesive pads (the Champ must go side-on). Two separate batteries are used to minimise interaction between the sections.**

**This is the EMM-6 calibrated microphone from Dayton Audio, which sells for about $80. Or you can make your own (as described in the text) for a whole lot less!**

by constructing the Pre-champ preamplifier as per the instructions given (or refer to the article in SILICON CHIP, July 1994).

Note that you need to change the values of three capacitors, as shown in Figs.1 & 2. These should easily fit on the PCB.

If all goes well, you can then start on the "Champ" power amplifier as per the kit instructions (or SILICON CHIP February 1994).

Again, there are slight modifications required. Figs.1 & 3 show these, which involve changing two capacitors. The 4,700µF capacitor does fit on the PCB but it is a bit too tall and the finished amplifier will have to be mounted on its side so it can easily fit in the diecast box.

Once the two PCBs are completed, you can drill and mount all the hardware on/in the diecast box using Fig.4 and the photos as a guide.

Solder all the connecting wires according to the diagram. It will be easier to solder the wires to the boards first then solder the wires to all the switches and sockets before mounting them inside the box.

Because the circuit boards are tiny and have no provision for normal screw mounts, you will have to use some good quality, thick, double sided foam pads.

Cut the pads to cover the bottom of the "pre-champ" board then press it firmly in place, allowing plenty of room for everything to clear.

The 100kΩ log pot is mounted directly to the diecast box for convenience but the original 10kΩ pot is retained on the PCB as a "preset" to take care of variations between sound card outputs. Later we'll set the maximum output of the Champ to prevent clipping and excessive distortion.

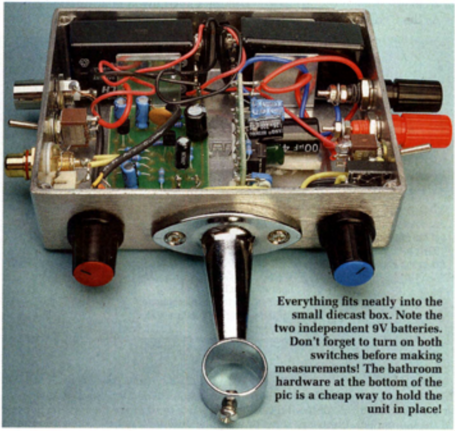
## The microphone

If you wish, you can make your own microphone to use with this system – details follow.

Or you can buy a ready-made calibrated microphone – for example, the EMM-6 Measurement Microphone from Dayton Audio (a company in Springboro, Ohio, USA) sells for about $US80. It's a precision electret condenser microphone designed for measurement and critical recording applications. However, this microphone requires a minimum 15V phantom power so you'll need to arrange a separate phantom supply (two 9V batteries in series would be fine).

Once you've purchased this mic you can then download its own calibration data text file.

Further information (including a



**Everything fits neatly into the small diecast box. Note the two independent 9V batteries. Don't forget to turn on both switches before making measurements! The bathroom hardware at the bottom of the pic is a cheap way to hold the unit in place!**

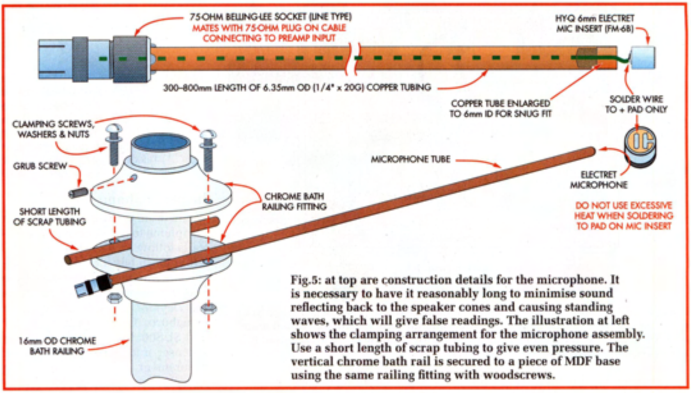



**Fig.5: at top are construction details for the microphone. It is necessary to have it reasonably long to minimise sound reflecting back to the speaker cones and causing standing waves, which will give false readings. The illustration at left shows the clamping arrangement for the microphone assembly. Use a short length of scrap tubing to give even pressure. The vertical chrome bath rail is secured to a piece of MDF base using the same railing fitting with woodscrews.**

spec sheet) is available from www.daytonaudio.com/index.php/emm-6-electret-measurement-microphone.html

## Making your own

You'll need a length of 6.35mm (1/4in) copper pipe, at least 300mm or so long.

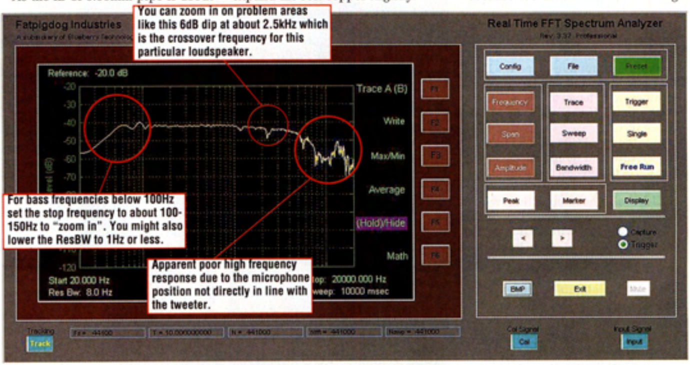
As the ID of 6.35mm pipe is about 4.85mm and the electret microphone OD is 6mm, you'll need to enlarge the end of the pipe to accommodate same, down to a depth of about 6mm.

Drilling the pipe out is possible but impractical due to the thin copper wall – it's much better to force a punch or something similar into the end to expand the soft copper slightly.

A pipe flaring tool might also be useful here but we haven't tried it.

Once done (check the electret fits but don't get it caught in the tube!), you need to solder a connection to it.

Using a clean, hot soldering iron, solder a single wire to the positive terminal of the electret – be careful because too much heat will damage




**The virtual analyser showing the frequency response of a three-way loudspeaker. You can adjust the start and stop frequencies to 20Hz-200Hz and resolution to 1Hz to improve the bass response curve. Note the tracking generator "button" at the bottom left. Insets are some things you could look out for when fine-tuning speakers.**

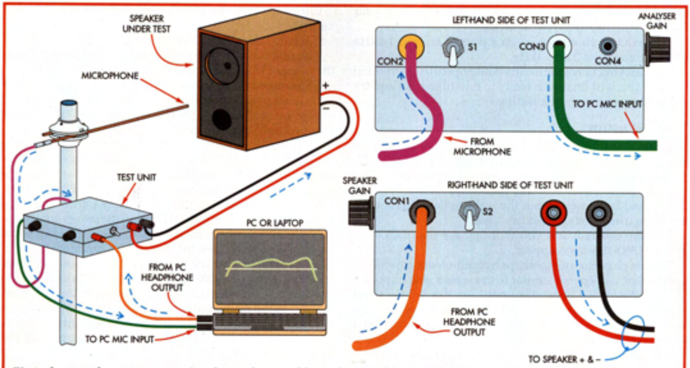



**Fig.6: the complete test setup using the analyser, calibrated microphone, modified amplifiers and software on a PC.**

the low-end response of the electret.

A gas powered soldering iron wound up fairly high is ideal.

(It is a good idea to buy two or three electrets in case an accident happens – they are quite cheap).

Then, run the wire down the centre of the copper tubing and mount a 75Ω female co-ax plug to the other end. The one we used required no solder and the wire was simply screwed into the centre then pushed back in.

The copper tubing then acts as the "ground" connector at both ends and also forms a good shield.

Cut a length of coaxial cable to about 1-2 metres long and fit a male co-ax plug to each end.

Once you have completed the microphone assembly, it is important to have a good solid stand so you can accurately position the microphone in front of the speaker under test.

We used 16mm bath rail fittings that you can buy from any hardware store.

We mounted a length to a piece of board, then clamped the copper tubing with two of the 16mm round ends using small nuts and bolts.

A "thru" chrome rail fitting was bolted to the diecast box and the opening was drilled and tapped to fit a clamping screw.

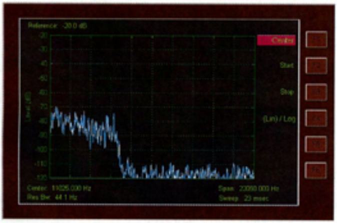
The alternative is to secure the unit by merely using insulation tape wound neatly around the vertical support to stop it from slipping down.

## Checking it out

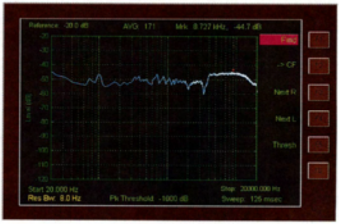
Now all you need to do is plug all the wires in as per Fig.6 and switch everything on.

Check to see if the microphone is working by talking or whistling and measure the output with a DMM set on AC (or plug the output into an amplifier or oscilloscope). The latter is best because you will see immediately if you are getting a clean sine wave.

Alternatively, you might like to plug the output of the Pre-champ into the



**Trace 1: this is how the vitual instrument should appear after startup in the factory-preset mode. It displays a resolution bandwidth of 44.1Hz, a sweep time of 23ms and a span of 22.05kHz (see Spyro's comments on how to set it up).**



**Trace 2: this looks like excessive bass but this is because the analyser and soundcard response is too slow with 125 milliseconds so we need to zoom in to the lower frequency range.**

mic socket of your computer soundcard and view your "whistle" on the spectrum analyser.

Your whistle should give you a peak at around 1-2kHz plus harmonics at 2 and 3kHz.

Once all your checks are done (and hopefully everything works!) you will finally be ready to fine-tune it all and try some frequency response testing.

## The test setup

We assume that you have downloaded the software from www.fatpigdog.com/SpectrumAnalyzer

The originator, Spyro Gumas, is very communicative and can assist if you have any problems.

We used Windows XP but the website lists alternatives for those using Vista, Windows 7 etc.

Run the program and you will first see the black-and-white MS-DOS screen appear.

You may have to wait (perhaps two minutes or so) and the instrument will appear similar to the screen grab opposite.

Once the virtual instrument pops up, this is how to set it up for frequency response measurements, making sure that the inputs and outputs to the test unit and computer are correct (see Fig.6).

Switch the test unit on and adjust the microphone so it is approx 40-100mm away, in a direct line, from the tweeter or speaker unit under test.

Connect the computer's headphone jack output to the input of the "Champ" power amplifier and attach the Champ output to the speaker under test (SUT).

(We converted the stereo output signal from the soundcard to mono at the input socket but one channel is OK).

On the virtual analyser:

Click on "preset" to clear any previous settings.

Click on frequency

Click on start (F2) and type in "20" <enter>

Click on stop key (F3) and type "20,000" <enter>

(The range is then 20Hz-20kHz)

Click on Lin/Log key (F4) so you see lin/(log).

The frequency range is set to a logarithmic scale 20Hz-20kHz.

Then:

Click on bandwidth

Click on RBW and type in "8" <enter>

Click on sweep, then click time (F2) and type "10000" <enter>

Click on "trace" and then "average"

The analyser will then sweep continuously and indicate the number of averages at the top of the page.

The analyser is now ready to do a 10-second sweep of your loudspeaker from 20Hz to 20kHz with a resolution of 8Hz and will average the response curve (5-50 averages will probably be sufficient).

Click on "track" and you should hear the signal sweep from 20Hz to 20kHz; this repeats every 10 seconds. You can adjust the volume of your loudspeaker as it sweeps and save an image anytime by pressing "BMP" (bitmap).

You may find that the low-frequency part of the trace jumps around. This is normal because the sweep is not slow enough (10 seconds is maximum) to allow the analyser to capture it properly (see traces 2 & 3 for examples).

To fix this, try starting the sweep at 20Hz and stopping it at 200Hz or even 100Hz, and play around with the RBW (resolution bandwidth), which you can set as low as 0.1Hz!

Refer to the manual (downloaded) if you have difficulty because some computers have different delay arrangements with the soundcard and you may need to compensate the analyser with Tstupid.

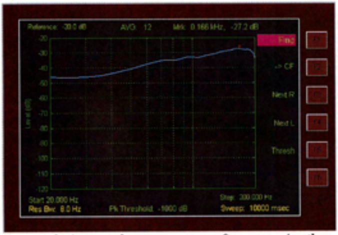
What is Tstupid? It's a part of the fatpigdog software. When data capture is initiated with the audio capture card in My PC, the initial gain response is zero, or pretty close to it. My audio card takes approximately 100ms for its recording gain to stabilise. Tstupid is an advance in the amount of time that the spectrum analyser captures data for a Single sweep or for the first sweep of a Free Run. The captured data during the Tstupid interval is discarded. The user has access to this parameter to use at his peril. The default value is 100.

You can also adjust the volume of the speaker and the gain from the microphone until you get a nice-looking trace.

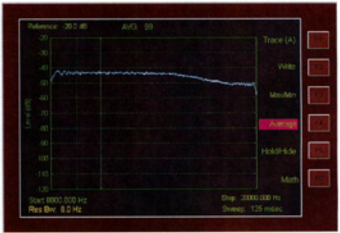
If you wish, you can make adjustments to your speaker while the analyser is sweeping; such as tweeter or midrange volume levels (if an L-pad is fitted) or by moving the microphone into different positions away from the tweeter.

When you are happy with a particular trace, you might like to activate the marker to examine a point of interest.

Click on "marker" then "ON" and you will see a red dot




**Trace 3: the improved response curve after narrowing the frequency range to 20-200Hz and keeping the 10s sweep time for 12 averages. You can reduce the Res Bw to 0.1Hz, but the analyser will take a longer time to do a trace.**




**Trace 4: narrowed to show 8kHz-20kHz response to zoom in on the tweeter. This speaker is very smooth but drops away 5dB or so at the higher frequencies. The dip at 20kHz is due to the microphone response being 2.75dB lower.**

appear on the trace

Then move the marker to the area you want to measure by clicking on "<" (backward) or ">" (forward) keys.

The marker reading appears at the top of the page eg, "Mrk 2.558kHz, -86.2dB"

## Correcting the microphone

Once you have measurements of the points you are interested in, go to the correction table below (Table 1) and add or subtract the dB value at the frequency of interest.

For example if you measured -26.5dB at 20Hz you have to **add** 3.7dB to get the corrected value because the microphone's own response falls off at low frequencies (see trace 5).

We aimed for an accuracy of ±1.5dB but by using the correction table we have achieved better than ±1dB.

**The measurement is dB relative to the reference signal. It is NOT a dB sound pressure level (dB SPL) measurement.**

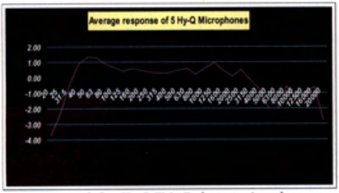
We cannot give you a reference because every soundcard will have a different internal gain.

To change to dB SPL you will need to calibrate your test setup against a known sound pressure level by using an accurate sound level meter or by using a "microphone calibrator" which emits a pre-determined audio output at point blank range

You may also use a speaker which has a specification for SPL, eg. 90db SPL at 1W 1m at 1kHz – but of course you

| Frequency (Hz) | ADD dB to measurement |
|---|---|
| 20 | 3.70 |
| 25 | 2.35 |
| 31.5 | 0.45 |
| 40 | -0.89 |
| 50 | -1.35 |
| 63 | -1.29 |
| 80 | -0.88 |
| 100 | -0.68 |
| 125 | -0.44 |
| 160 | -0.60 |
| 200 | -0.46 |
| 250 | -0.33 |
| 315 | -0.28 |
| 400 | -0.31 |
| 500 | -0.47 |
| 630 | -0.59 |
| 800 | -0.23 |
| 1000 | -0.59 |
| 1250 | -0.96 |
| 1600 | -0.47 |
| 2000 | -0.08 |
| 2500 | -0.48 |
| 3150 | 0.16 |
| 4000 | 0.78 |
| 5000 | 2.02 |
| 6300 | 2.02 |
| 8000 | 0.57 |
| 10000 | 1.33 |
| 12500 | 0.99 |
| 16000 | 0.64 |
| 20000 | 2.75 |

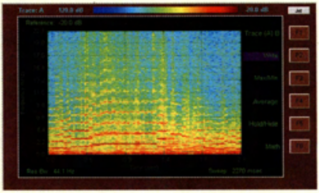
**Table 1: correction table for HY-Q FM 6B electret microphone.**




**Trace 5: we took five Hy-Q FM-6B electret microphones and averaged their responses at a range of frequencies to produce the curve above. The same figures are reproduced in table form above. Using these figures you can correct for variations in the microphone response. For example add 3.7dB to your reading for 20Hz and 2.35dB to the 25Hz reading and so on. Accuracy after correction will be ±1.0dB.**

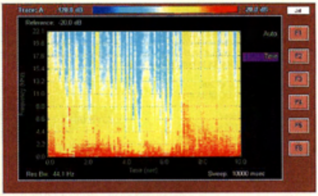
# Other Applications

The software will also enable you to do waterfall analysis. This is really a way of viewing a spectrum analysis as it varies over time. It can be used for making "voice prints" or charts of audio signals.



**The instrument also does waterfall charts in beautiful colours with frequency (horizontal axis) vs time (vertical axis). Colour code is at top and represents the intensity of the signal.**

The screen grab above shows the waterfall chart for 2.270 seconds of the Bruch Concerto No 3 for violin and shows the rich harmonics. The vertical scale shows the frequencies of the various harmonics while the horizontal scale is time so the whole chart is a record of a few notes of music.



**A waterfall of Shakira singing "How do you do". Interpretation of these charts is strictly up to your imagination!**

## To set up for Waterfall Charts

The wiring setup is virtually the same as for testing loudspeakers except that music or voice has to be fed to the loudspeaker from a CD player or MP3 player, or from the microphone "Pre-champ" output (for voice prints)

The setup for the virtual instrument is:

Click on "preset"

Then "display"

Then "waterfall F2"

Then "rotate"

Then try different sweep times and resolution bandwidths (Res. Bw…).

And try different colour schemes by clicking on "jet"

Press BMP to save the image you want.

will need to push the champ to 1/2 watt (ie, 2V RMS for an 8Ω speaker) at 1kHz by

clicking on "frequency"
then "centre frequency'
then "1,000 enter"
then "span"
then "zero F3"
then "track"

This will now set the generator at 1kHz and you can feed this to your speaker (you will hear a clicking sound on each sweep so set the sweep time to 10,000mS).

The real SPL at 1m will then be close to an SPL of 84dB (1/2 the specified value) or 90dB at 0.5m (because watts=$V^2R$ and sound level is an inverse square function).

That is only true if the manufacturer's specification is correct, so you might try different speakers – or just don't worry about it if you don't really need it!

## Preventing clipping and distortion

You can set the maximum output from the "Champ" by setting the preset at a value which prevents clipping and excessive distortion.

You can do this by setting the spectrum analyser centre frequency to 100Hz and then "zero span".

The maximum output to the speakers can then be measured with an AC voltmeter (make sure you fit an 8Ω, 0.5W resistor as a dummy load) and the preset adjusted so the output does not exceed 1.5V RMS and that you have fully advanced the 100kΩ pot. Once this is done, you can be certain that you will not accidentally clip and distort the signal going to the speaker. SC

## Parts list – Speaker Testing

1 Diecast case, 119 x 94 x 34mm (eg Jaycar HB5067)
1 "Champ" amplifier kit (Silicon Chip, February 1994)
1 "Prechamp" preamplifier kit (Silicon Chip, July 1994)
1 6mm electret microphone insert (Hy-Q Electronics FM-6B)
2 SPST switches (panel mounting, any type)
1 75Ω panel socket
1 75Ω male plug
1 75Ω line socket
1 banana socket (black)
1 banana socket (red)
1 RCA socket
1 3.5mm stereo socket
1 3.5mm mono socket
1 length coax cable (~1m)
2 knobs (colours to suit)

**Note: nominated parts were those used in the prototype but you can use plugs/ sockets etc you may have on hand.**

**Capacitors** (changes to components supplied in kits)
1 4700µF 16V electrolytic
1 470µF 16V electrolytic
1 4.7µF 16V electrolytic (non polarised preferred)

**Potentiometers**
2 100kΩ miniature panel mount type

**Software**
Fatpigdog Virtual Analyzer (see text)

**Hardware**
1 length 6.5mm x 20G annealed copper pipe (~500mm)
16mm chrome bathroom fittings as required

## *A word from Spyro Gumas, originator of the Fatpigdog Spectrum Analyser*

The inspiration for the name "Fatpigdog" is our pug Buddy, a truly Fat Pig Dog. The inspiration for the software itself was my frustration in trying to use virtual spectrum analysers with their non-intuitive user interfaces. Having used spectrum analysers quite a bit, I yearned for a virtual tool that worked the same way the real hardware tools work. I can't say I've totally achieved this objective but I do think that anyone with experience using an HP, Agilent or Tektronix analyser will find my software so easy to use that they can throw away the Users Manual.

The spectrum analyzer starts up in a factory preset mode, displaying the full frequency (SPAN), with an update time (SWEEP) of 23ms and a Frequency Resolution (BANDWIDTH) of 44.1Hz. This will get you started, but lets say that you decide to drill a little deeper. You're playing with Ye Olde Fatpigdog Spectrum Analyser (that's how we all talk up here in the states) while watching your favorite television program on your old fashioned (tube) TV.

You notice a strong signal peak centered at 15.734kHz (NTSC system, 15.625kHz for most of you other folks) and wonder if that could be the arcane horizontal sync frequency emanating from the sync oscillator.

So, you click FREQUENCY, type in 15734 (humor me) for the center frequency and hit Enter. So far so good, the display has shifted, but now you want to adjust the span so you can zoom in on any possible spectral structure. So, you click SPAN, type in 100, and click Enter.

Whoa, everything comes to a crashing halt. The display is now updating once every 5 seconds. Why?

So here's the secret. With SWEEP and BANDWIDTH in the default AUTO modes, the spectrum analyser is going to automatically set bandwidth equal to SPAN/500 [This ratio is a magic number that you can change under the CONFIG menu, labeled Span/RBW.] Now here's the science behind Resolution Bandwidth (RBW): to get frequency detail at a resolution of RBW Hz, you need to analyse a length of audio signal that is 1/RBW seconds long. So when we set our SPAN to 100Hz, the spectrum analyser automatically set RBW to 0.2Hz (100Hz/500) and then computed a corresponding SWEEP time of 5 seconds (1/0.2Hz). Aha.

So what can you do about this? ... A Lot! Don't let the software push you around. You've been given full flexibility, courtesy of the wizards at Fatpigdog Industries. You can change the magic number Span/RBW to something like 50 and voila, the SWEEP goes to 500ms. But this is kind of gross, to be truthful since the frequency resolution is very coarse now. So, let's set Span/RBW back to 500. Now click SWEEP, and then the TIME soft key. Enter 50. Now the Sweep is updating every 50ms, but the bandwidth is still very fine (RBW still is 0.2Hz).

But it looks strange, a certain squirreliness to it. That's because the spectrum analyser is still processing 5s blocks of data to generate the fine frequency resolution but its processing a sliding 5s window of data, every 50ms. This means that every 50ms it is processing 50ms of new data and a residual 4950ms of data from the last update.

Thus you are seeing fast updates, but the spectrum is the result of averaging over 5s. It's a compromise! That's how it works, you trade off speed for frequency resolution but you can get both if you are willing to smear the spectral changes over time.

I like to think of this as the time/frequency Heisenberg Uncertainty Principle ... more on that some other time (but you certainly can Google it!). I hope you enjoy the Spectrum Analyzer.