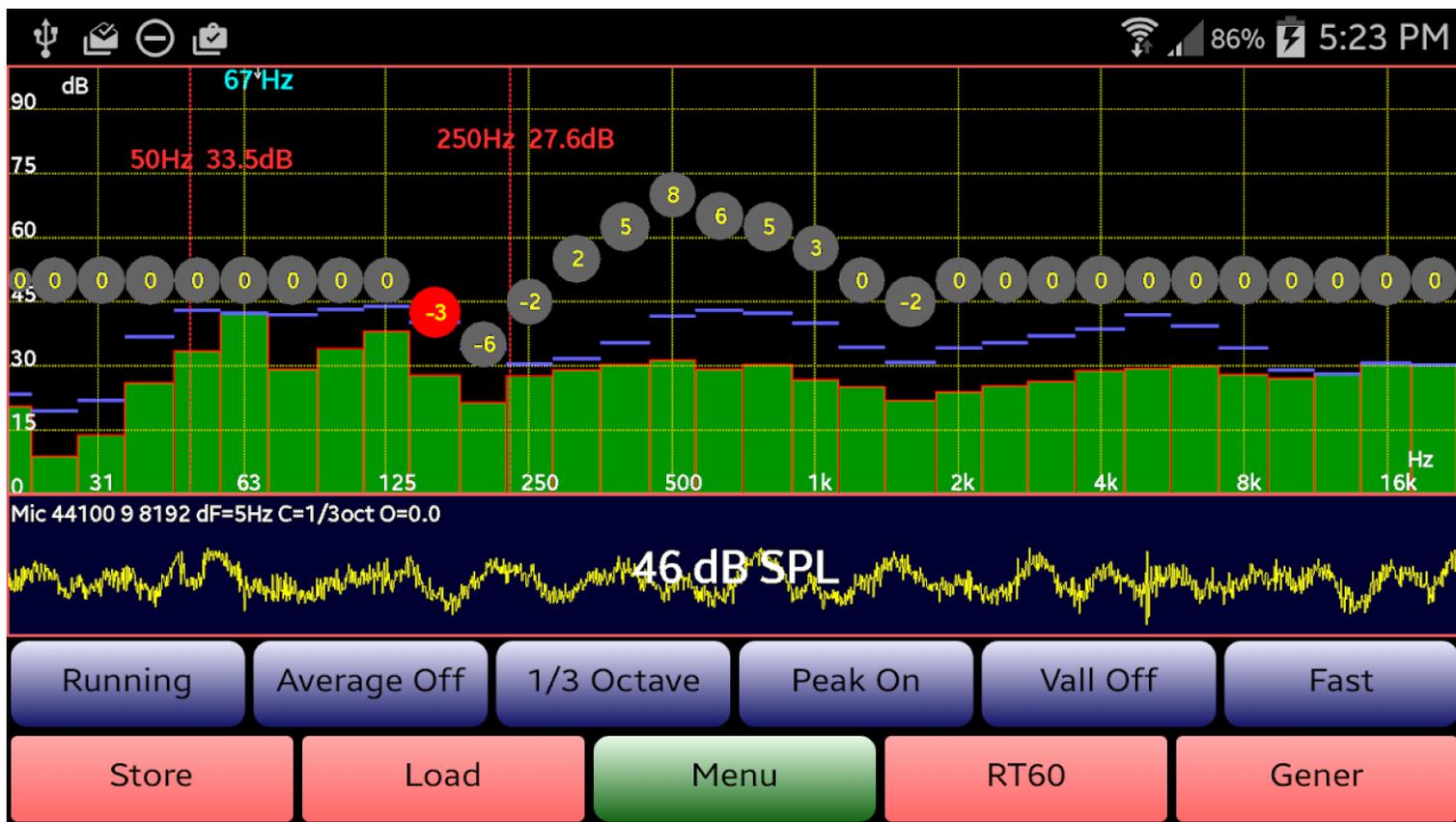


# AudioTool - Online Manual



# PDF Version

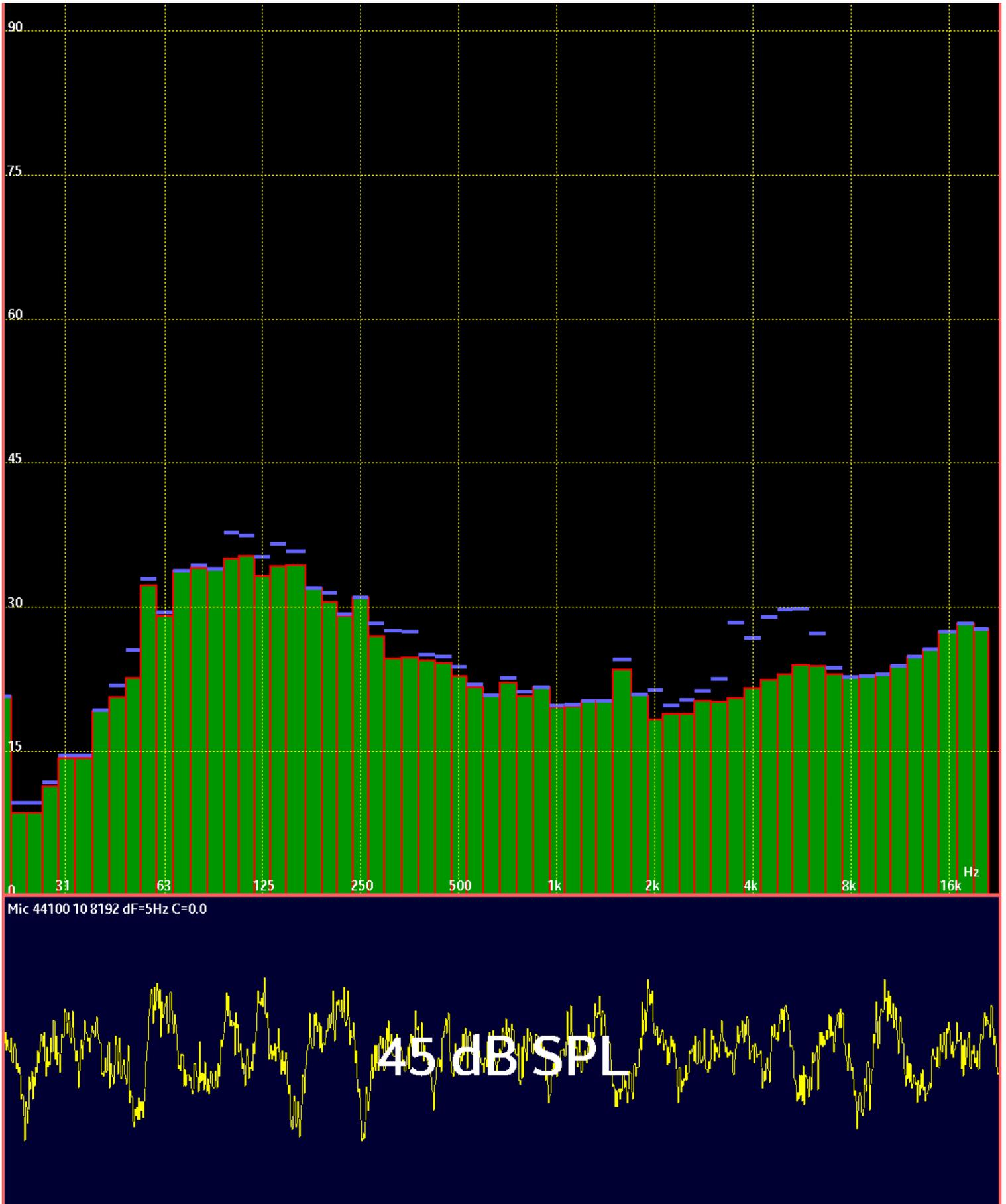
A PDF version of this online manual is available [AudioTool PDF](#) .

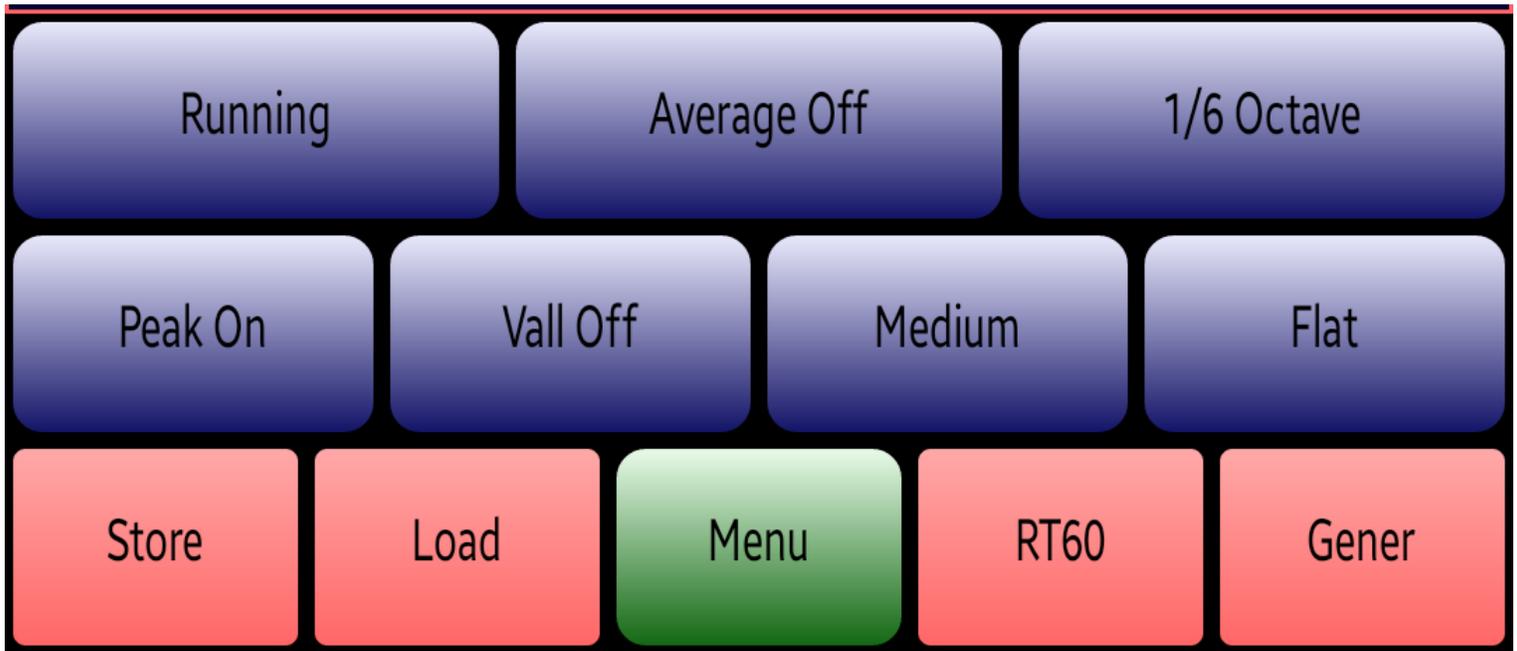
## Main Screen

When AudioTool is started, you will see the main screen for the Real Time Analyser (RTA). The screen is divided into three regions:

- The top region shows the RTA frequency response in ISO one octave bars, as calculated by an FFT of the incoming audio data. The vertical axis is labelled in deciBels. The horizontal axis is a logarithmic frequency scale that is marked at the ISO frequency centres (unless the user has selected the linear frequency scale option, see below). The height of the bars changes with the incoming audio signal being picked up by the microphone (or by an input device connected to the phone's headset jack). Also shown is the current peak frequency, in Hz (the screen shot shown has 121Hz) - this value is typically accurate to within a few Hz, depending on the sampling rate (see below). Also shown above each of the octave bars are blue marker lines that indicate the maximum level observed so far at that frequency.
- The middle region shows an oscilloscope-like realtime display of the audio data being picked up by the microphone. The vertical scale dynamically adjusts to the prevailing levels. Also shown in large text in this region is the currently measured SPL level, as calculated directly from the audio data stream (this number is not FFT based.) In smaller text, at the top left of the region, are three numbers following the word "Mic". These numbers indicate:
  1. The sample rate, in samples/second (44100 in the example)
  2. The "frame time" for the FFT in milliseconds (i.e. how quickly your phone is calculating - lower numbers are better)
  3. The sample buffer size being used for FFT analysis (8192 in this example).
- You can hide the middle (Input) region from view, using the "Hide Input" menu option. This results in more screen real estate for the RTA display.
- The lower region is occupied by control buttons, described below.







## Button Controls

The labels on the buttons indicate the current state of the control.

- **"Running" "Paused"** - this button is used to pause or run the data analysis. When "Paused", the RTA display does not change. The middle region 'scope display, however, continues to show the incoming audio.
- **"Average Off" "Averaging"** - for averaging the RTA analysis. When activated, the RTA levels will be averaged continuously. The count of samples averaged appears in the top, RTA region. Averaging is arithmetic, not power.
- **"1/1 Octave" "1/3 Octave" "1/6 Octave" "1/12 Octave" "Full Res" "Spectrogram" "SPL" "Leq" "Chart Recorder" "Peak Store"** - this is the button used to switch between the various modes available in the RTA, each is explained further in the next section.
- **"Peak On" "Peak Off"** - this controls whether peak levels in the RTA display are shown or not. To reset the levels is simply a matter of pressing this button to Off then On again.
- **"Valley On" "Valley Off"** - controls whether valley levels (lowest levels) are shown. Reset by turning off then on again.
- **"Medium" "Fast" "Slow" "Impulse"** - this control selects the speed of the RTA bars and the dB SPL measurement. Exponential filters are used with time constants of 0.5, 0.125, 1.0 and 0.025 seconds, respectively.
- **"Flat" "A Weight" "C Weight" "X Curve"** - this control selects between no filter (flat) and A and C

weight and the film industry X Curve filters, which are applied to the raw audio data and thus affect the RTA levels and the dB SPL level.

The bottom row of buttons, select major functions of AudioTool. The middle button is the Menu button. The Menu can also be activated by a long press on the phone/tablet's "Recent Apps" button.

To **hide the buttons from the screen** simply tap in the top region of the display. To **restore the buttons**, tap anywhere in the display.

## Microphone Select

An option in the Menu allows to select/toggle between the front microphone, for those phones that have one, and the main microphone (usually at the base of the phone). The front microphone is the one that is typically on the front of the phone used for video recording. The setting is remembered between sessions. For phones that do not have a front microphone, then the menu option is still available, but all it will do is select the main microphone's AGC stream - which is probably not what you want :-)

## Connecting an external Microphone/Amplifier

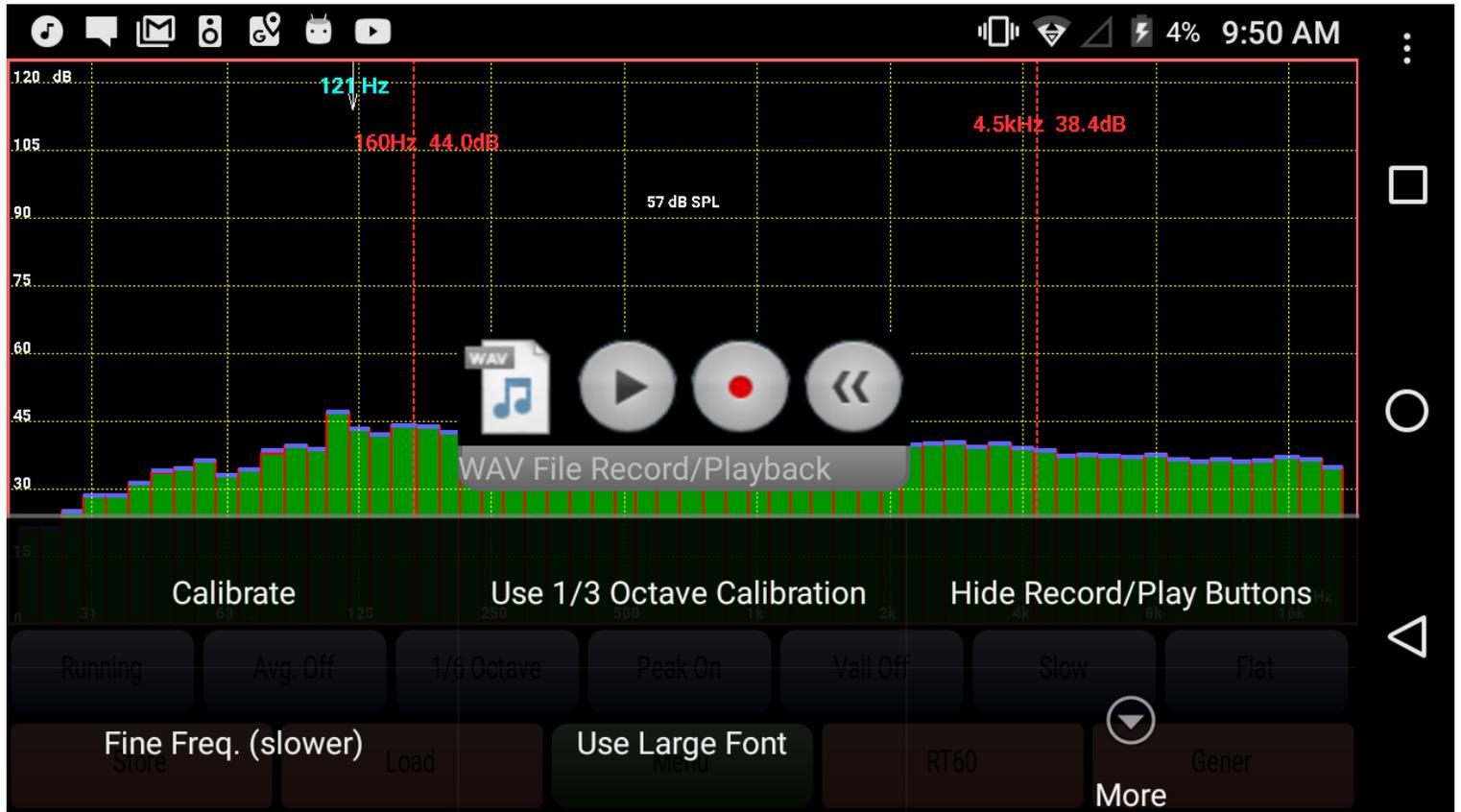
To connect an external microphone (the Dayton Audio iMM6 is recommended), just plug the mic into the headset jack on your Android tablet/phone.

To connect your amp, you can plug a cable with a jack on one end (into the iMM6's socket for that purpose) and phono plugs, or whatever your amp needs, on the other. Then, you can send signal generator audio to your amp, and measure the resulting audio with the iMM6.

If you don't have an iMM6 then you can use a Y connector in the headset jack - this breaks the jack into two jack sockets: one for a mic, one for audio out.

## Recording and Play Back

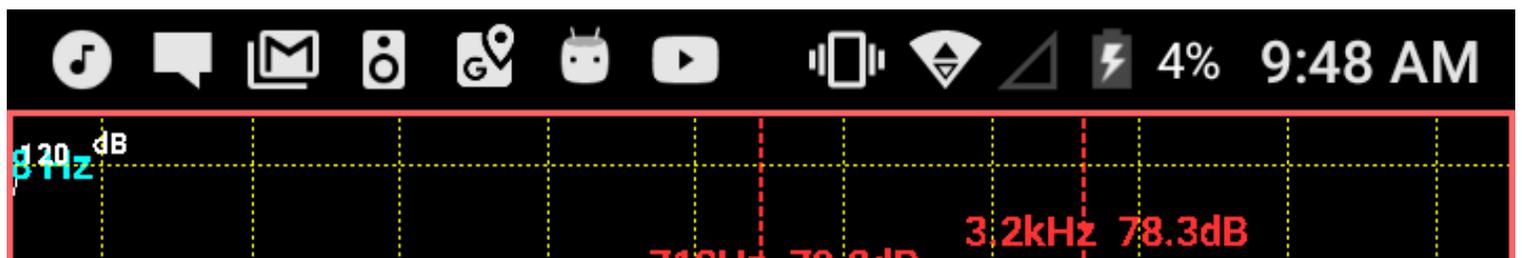
You can record audio samples from the microphone (internal or external) using the Record/Play button controls. To show the button controls, select the "Show Record/Play Buttons" item in the Menu. To hide them, select the "Hide Record/Play Buttons" in the Menu.

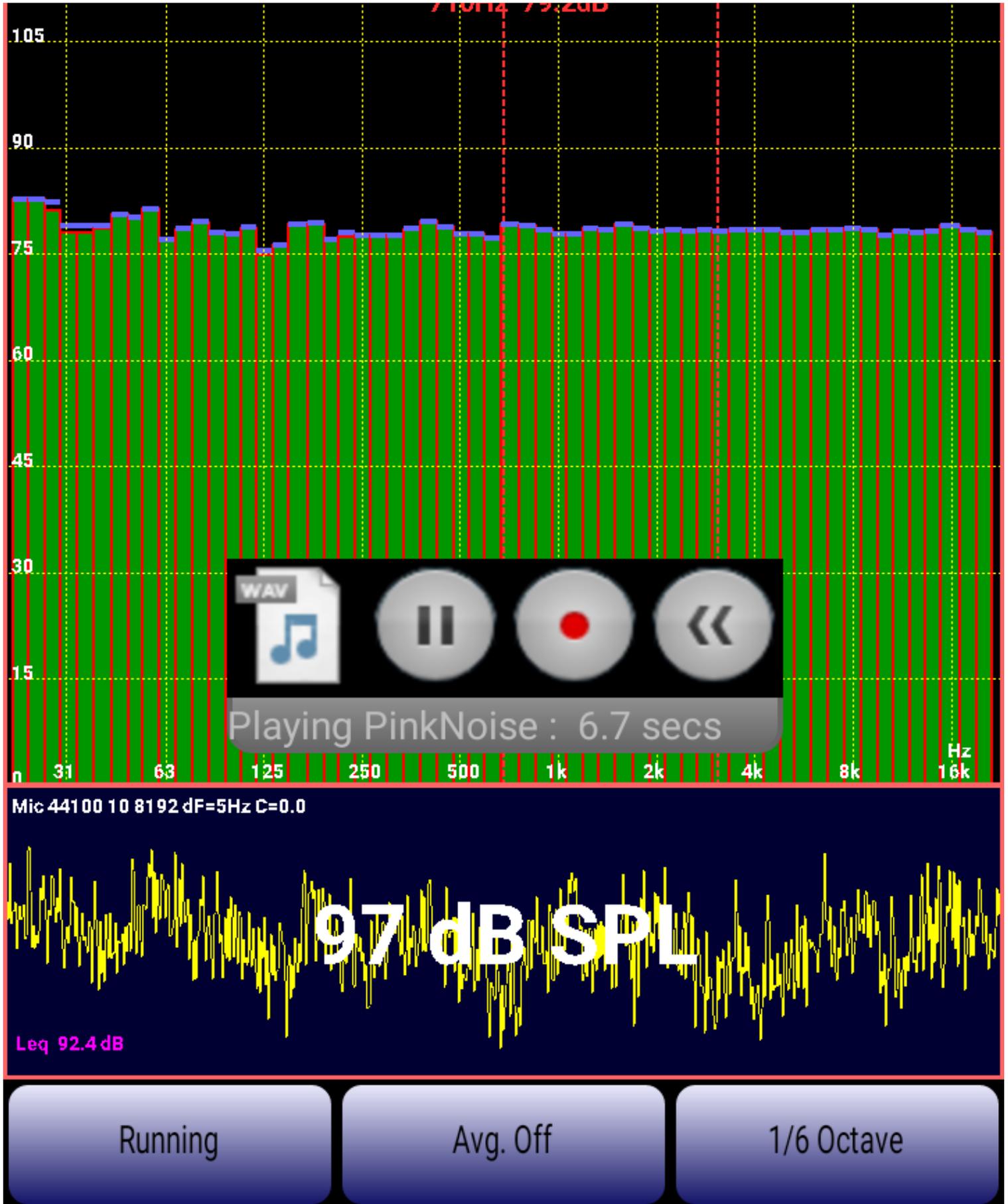


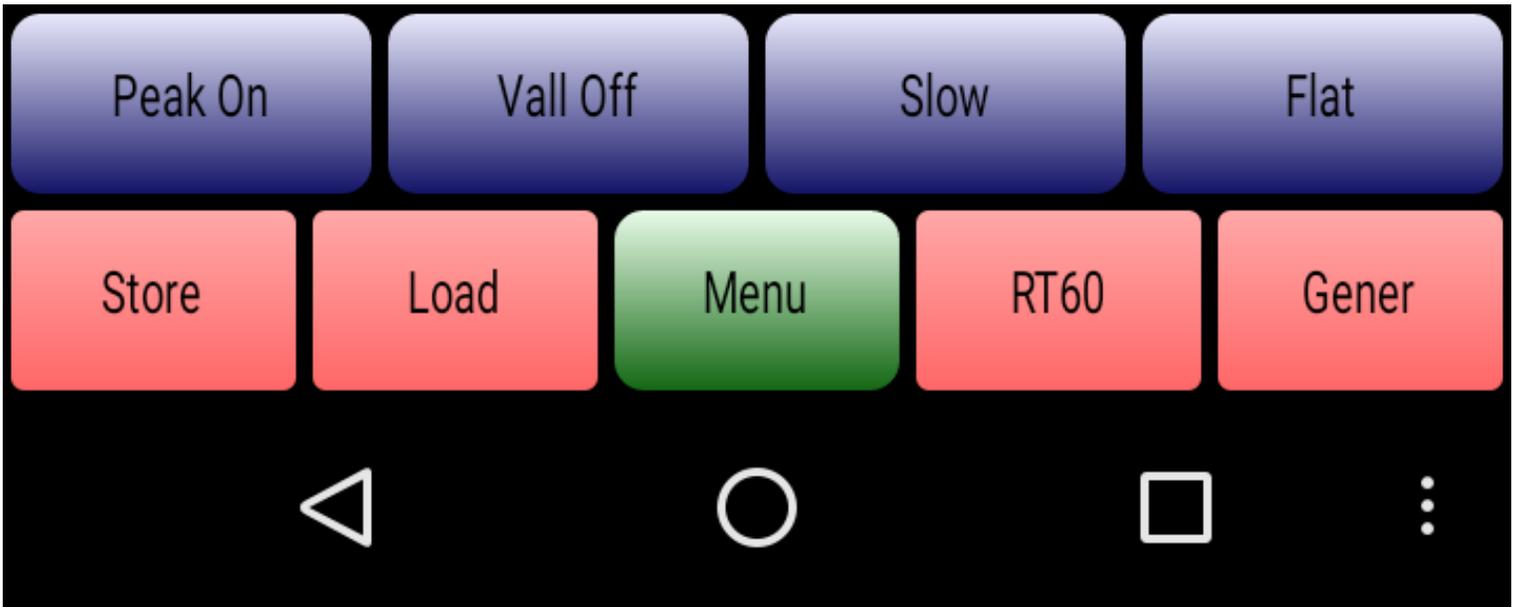
There are four buttons available:

1. WAV file selector - pressing this button will show the available WAV files in the AudioTool directory.
2. Play/Pause button - will play back samples from the WAV file selected in 1. Pressing this button during playback will pause playback.
3. Record button - press this button to start recording from the microphone. Press again to stop recording. A maximum of 30 seconds of audio data can be recorded. When finishing recording, a dialog will allow you to name the recorded WAV file, or discard it.
4. Rewind button - when pressed during WAV playback, the playback position will jump back to the start of the WAV file.

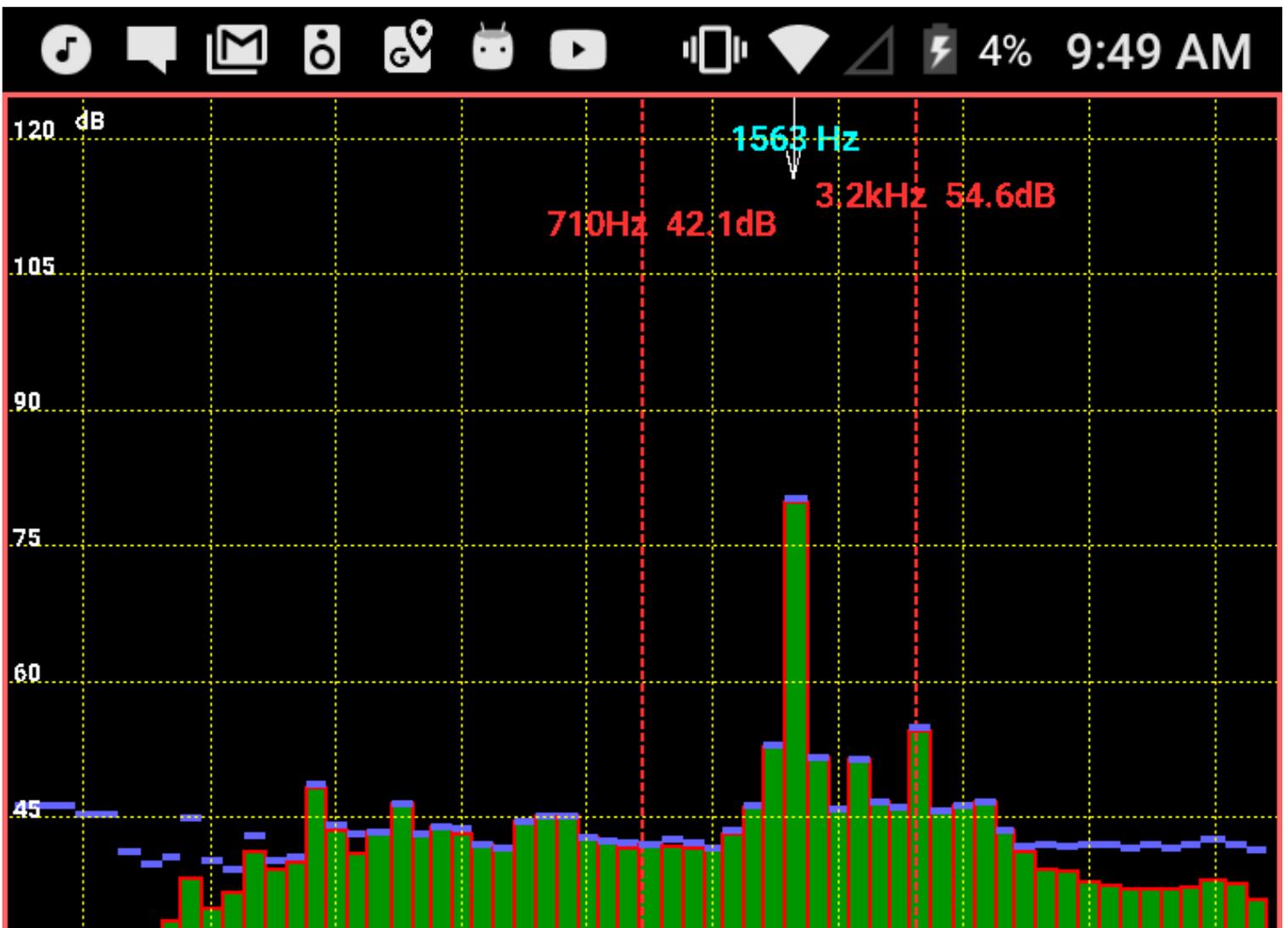
Here is an example where a WAV file called "PinkNoise" is being played back.

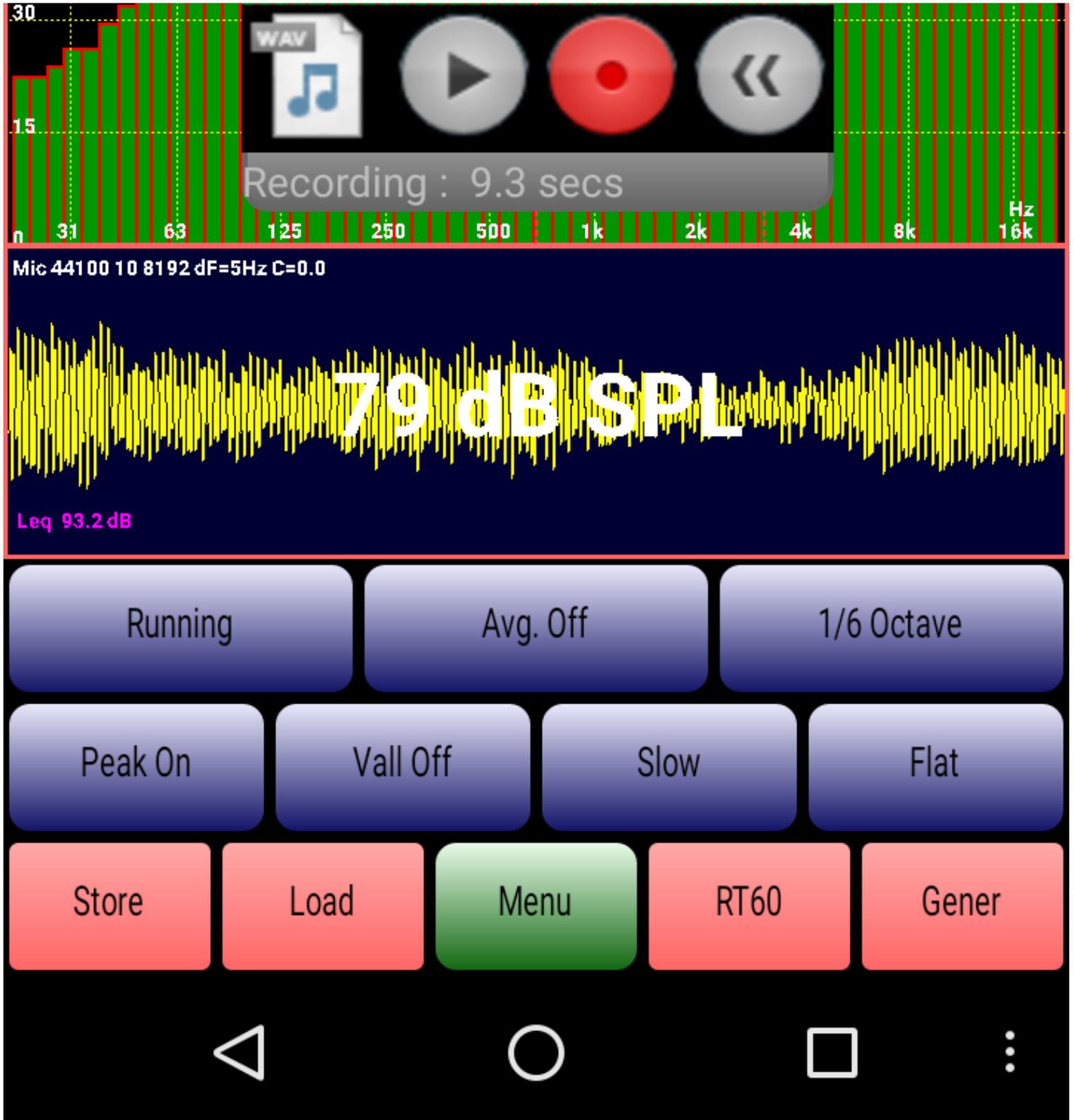






Here is an example where a WAV file is being recorded from the micriphone:



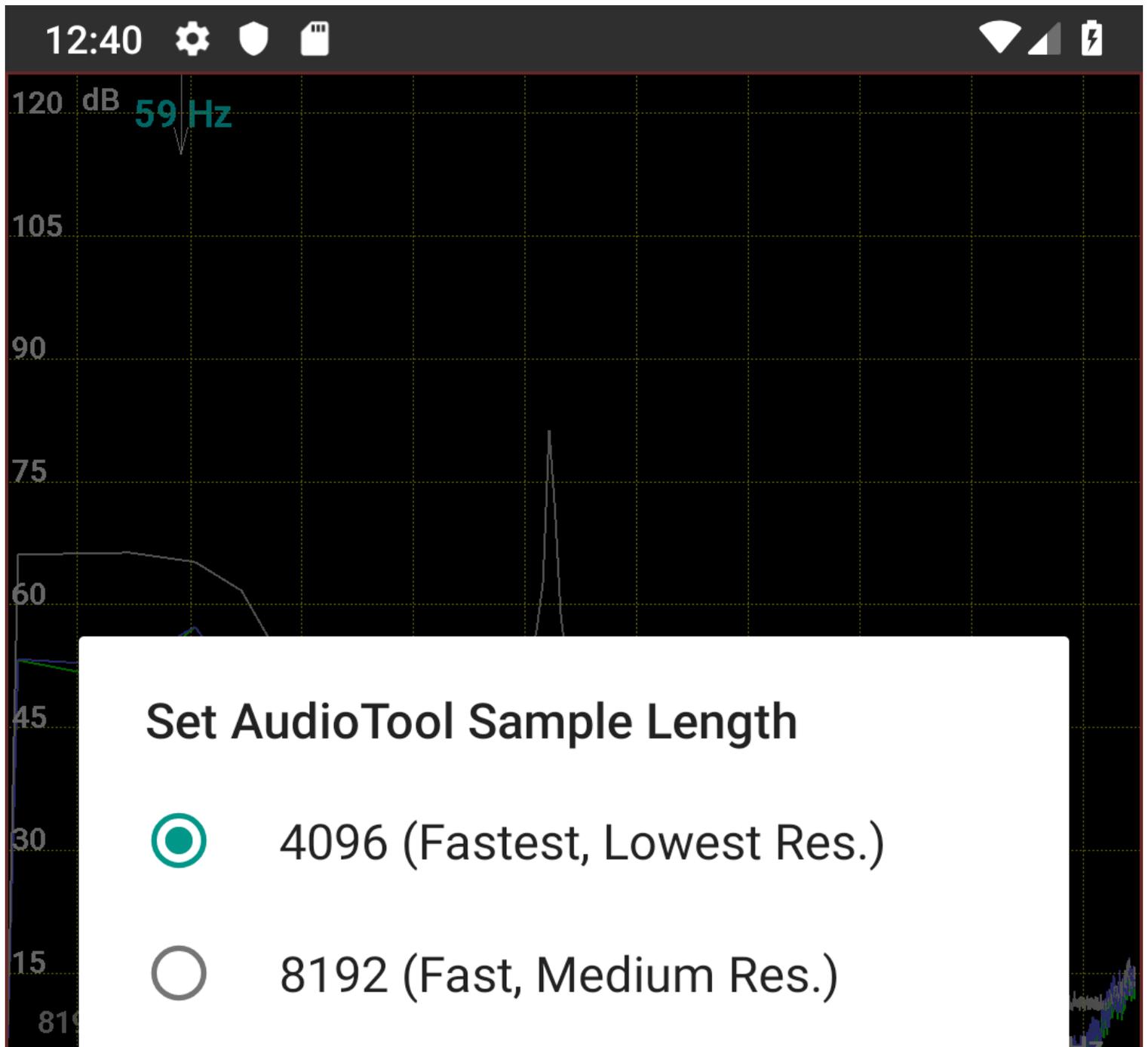


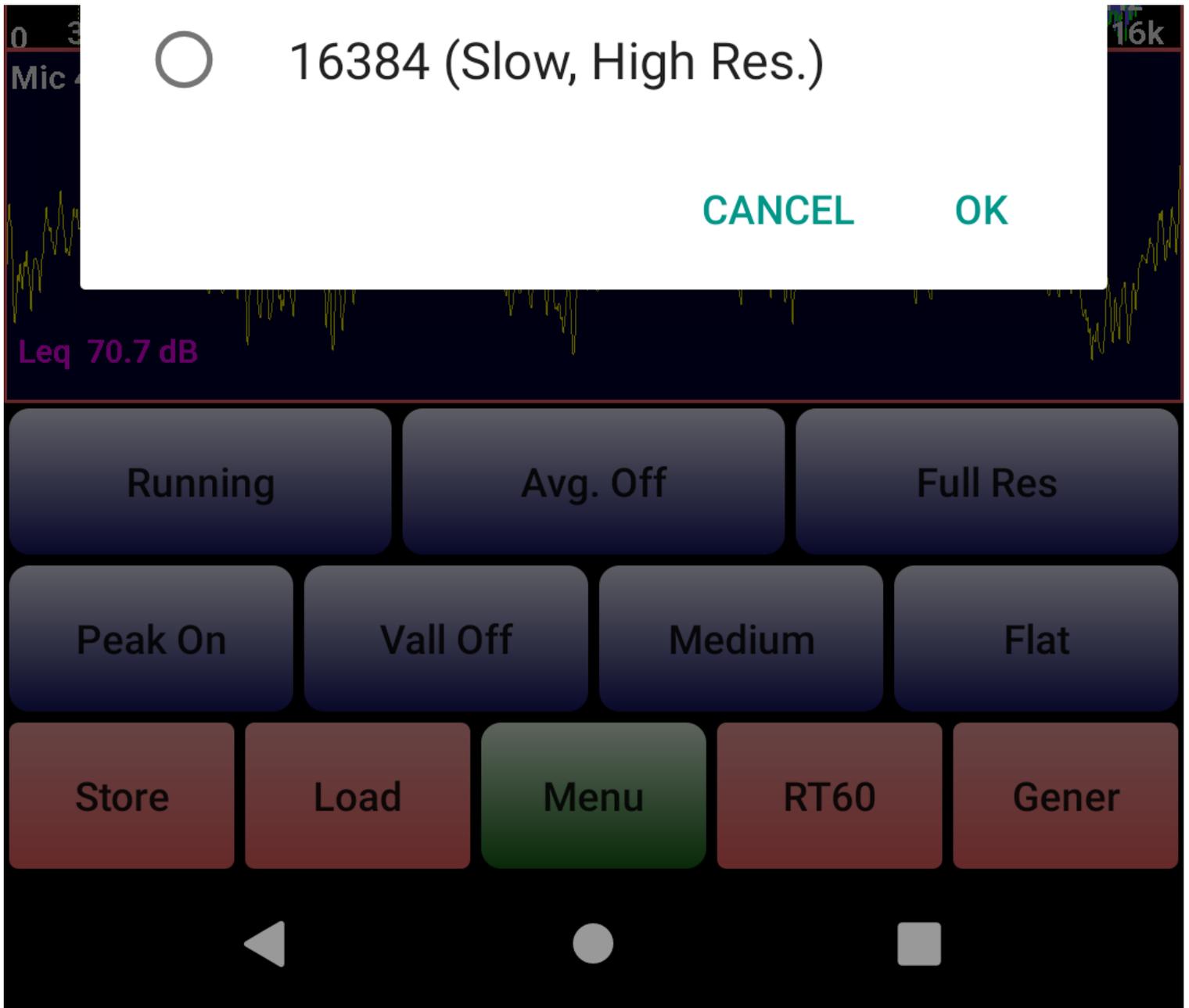
You can use any WAV file for playback, as long as it is recorded at the same sample rate as AudioTool is operating at: typically 44100 samples/second.

## Setting a longer Sample Length

New in v8.2

You can choose from three sample lengths using the "Set Resolution" Menu option:





The frequency resolution of AudioTool is governed by the sample rate (typically 44100 samples/sec) and sample length by the relation:

Frequency resolution = Sample Rate / Sample Length

So, the resolution at 4096 sample length is around 10Hz. The downside of increasing the sample length is that the frame rate will slow down (the display will not react as quickly as before). The advantage is that the frequency resolution will increase: this will be very noticeable in the RTA "Full Res." mode, for example.

# Store Button

This allows you to store the current RTA analysis, Chart, Leq, Peak Store or RT60 data into a file in memory. A dialog box will appear in which you can enter the name of the file in which the data will be stored. These files are written to the "AudioTool" directory in memory, and have a file type of ".at", ".rt60", "leq" ".cht" or ".swp" depending on which mode you are in. They are text files and can be read by any text editor.

## File Format for AudioTool .at Files

The AudioTool .at files have the following format. There is a file header line, containing the date/time etc., followed by rows of measurement data. For each row:

- The **first** data column is the frequency, in Hz.
- The **second** column is the magnitude of the FFT at that frequency I.e. it is the uncorrected voltage level.
- Previous to v6.0.3: The **third** column is a little more complicated: it's the square root of the corrected dB level, and thus depends on the current calibration, and the correction for the Von Hann window. This column is only present for historical reasons
- From v6.0.3 on: The **third** column is the corrected dB level. (It takes into account the current calibration, and the correction for the Von Hann window.)
- The **fourth** column shows the Peak dB value measured at that frequency
- The **fifth** column shows the Valley (minimum) dB value measured at that frequency

If you are in 1 Octave, 1/3 Octave or 1/6 Octave mode when Storing the file, the relevant octave data will be appended to the data described above. An example is shown below:

1/3 Octave Data: Columns are Frequency(Hz), dB, dB(valley), dB(peak)

20.0 26.83156706582885 22.166787511483037 52.632521644880924

25.0 24.79307357192959 22.166787511483037 52.632521644880924

31.5 23.638651696759617 14.802948453138242 54.36133378082017

40.0 31.207812167904592 19.370791325960823 45.49667680795788

50.0 38.912702880111574 29.64552758219944 49.49161872432127

63.0 31.308559430337557 22.89743713538305 48.84070943868894  
80.0 36.503730339760416 30.2550007563 53.60647538837455  
100.0 37.32872961438841 31.628594404178614 52.84494589661469  
125.0 47.53906365527617 41.49260507497887 55.603996089423774  
160.0 37.13218036161636 30.019891483037632 53.51520224280762  
200.0 38.61583915476531 32.18272210005908 56.303036331405195  
250.0 39.45571992076893 34.41176902389282 58.93353652741962  
315.0 35.445424637925015 32.892587984824274 53.20546791705307  
400.0 38.07353309291713 33.10039544690382 58.651517195975586  
500.0 37.44883468905982 31.62128553573676 60.243290257242265  
630.0 33.905814941577894 29.745581687854823 61.621031458186906  
800.0 31.189947485876157 29.136970289841344 62.9863347566566  
1000.0 29.934049998698615 26.281615541776937 68.23173512416146  
1250.0 28.804920066416592 25.35837631687501 72.65160623462893  
1600.0 29.091712814182326 25.780246296745275 92.86700351806023  
2000.0 27.885411613466054 24.668718676082374 77.11216044766489  
2500.0 27.272514725191616 24.29759590065956 69.88265084063536  
3150.0 27.044774522628643 24.289355987064475 70.62307656703035  
4000.0 27.96804350430325 25.149385241068856 65.93333352809589  
5000.0 28.201004393959373 25.481078088048356 62.52033892733458  
6300.0 33.39460194392798 29.18232982337404 63.791460855372634  
8000.0 31.42957001418992 28.356561247922514 59.74561820466221

10000.0 29.048496309608893 26.7129789759191 55.35978669914826

12500.0 29.135908066371623 27.80860720298393 54.31129910809918

16000.0 30.411527191811615 29.062520150477052 54.33884829589135

20000.0 28.65506503472534 27.471865999372465 53.57466153606412

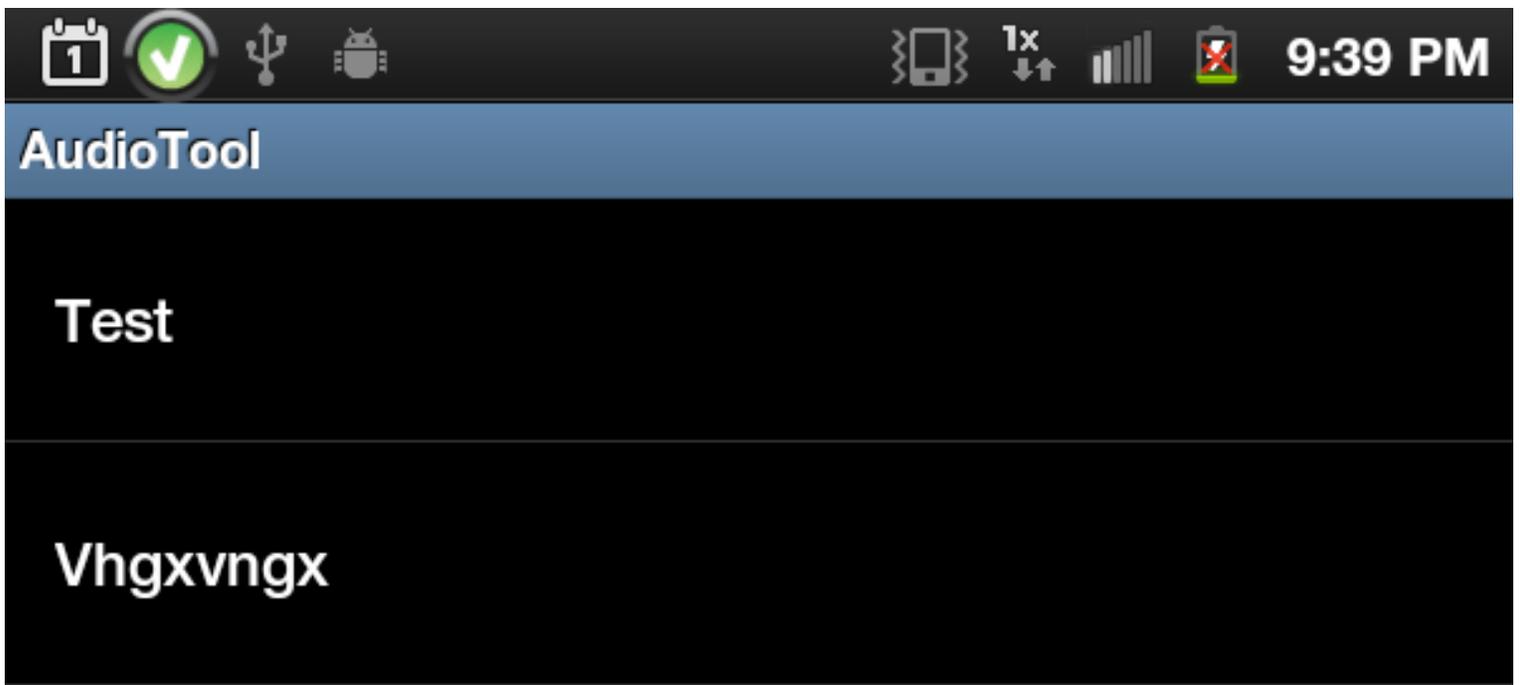
Note: for versions prior to v6.0.3 you can calculate the third column from the second (call it X) using the formula:

$$\text{Sqrt}(\text{dBcorrection} + \text{dBcalibration} + \text{dBvonHann} + 10\log(X*X))$$

Where dBcorrection is 20, dBvonHann is 4.2 and dBcalibration is shown in the file header. For v6.0.3 and above, remove the sqrt in the formula above.

## Load Button

This allows you to load up to two previously saved AudioTool .at files - they will be displayed alongside the real time data. This button is greyed out if there are no existing files of the appropriate type found in memory. If there are, a list will be shown of available files, together with two other options:

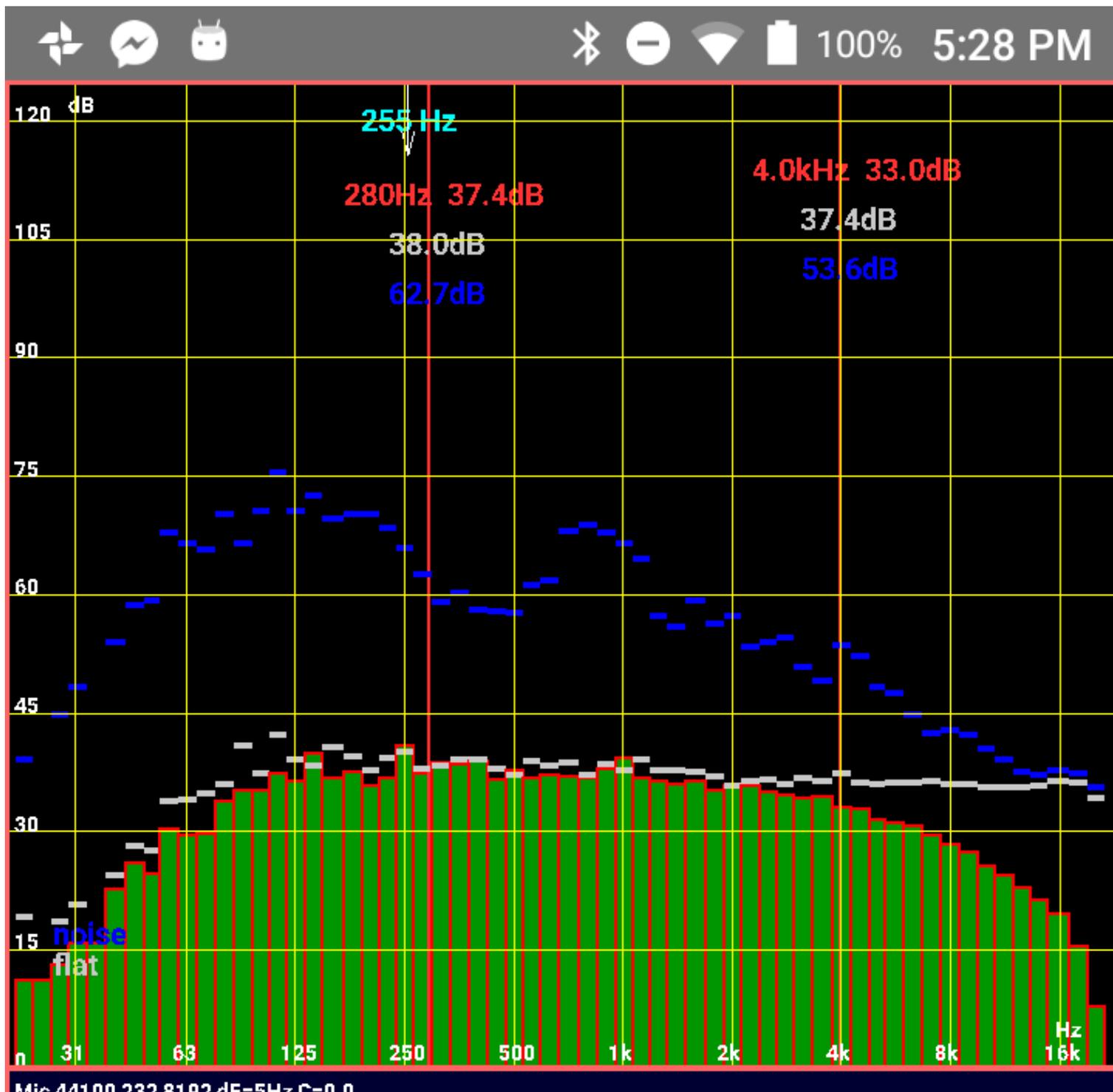


# Testing

<CANCEL>

<CLEAR>

Selecting <CANCEL> will return you to the main display, selecting <CLEAR> will remove an existing loaded file from being displayed. Selecting one of the files results in it being shown on the main display. You can select a second file to be Loaded by using the Load button again. Here is an example where two files have been loaded, one called "noise", the other "flat":





If you have enabled Cursors (see later in the manual), then the levels of any loaded file at the cursor positions will be shown. In the above example, the left red cursor shows that the "noise" curve (shown in grey) has a level of 38.0dB at 280Hz, and the "flat" curve (shown in blue) has a level 62.7dB at the same frequency.

## Menu Button

Tap this to reveal the Menu. Select the required option from the scroll list, or tap Cancel (bottom right).

## RT60 Button

This starts the RT60 feature - see below.

## Generator Button

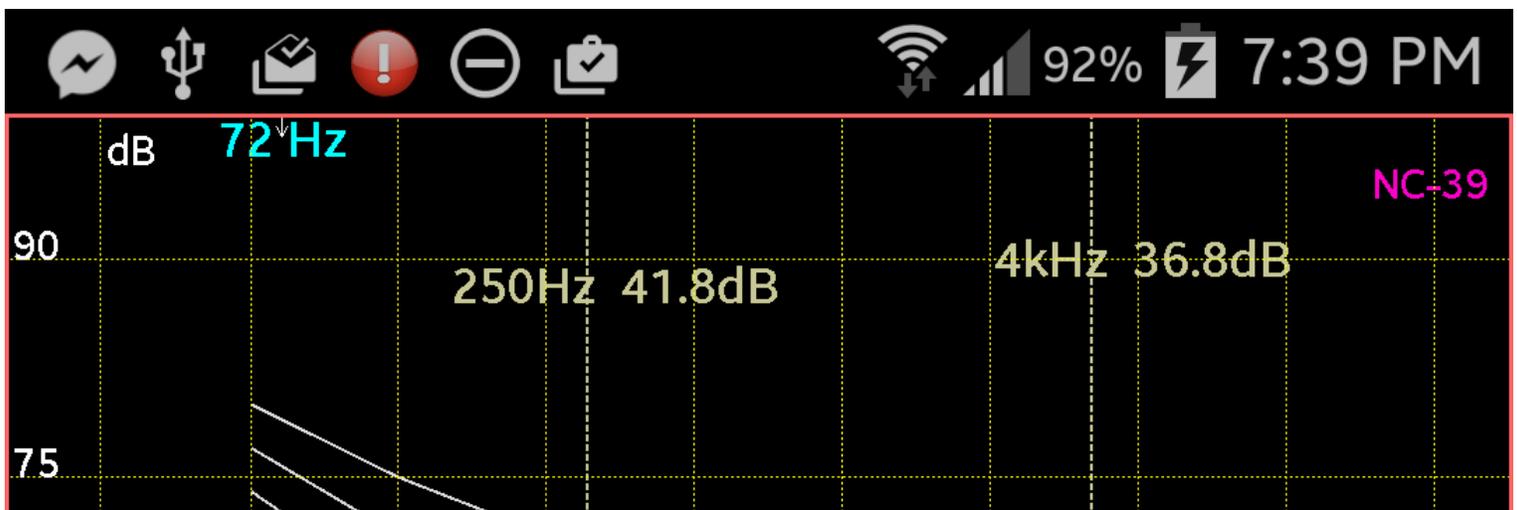
This starts the signal generator - see below.

## RTA

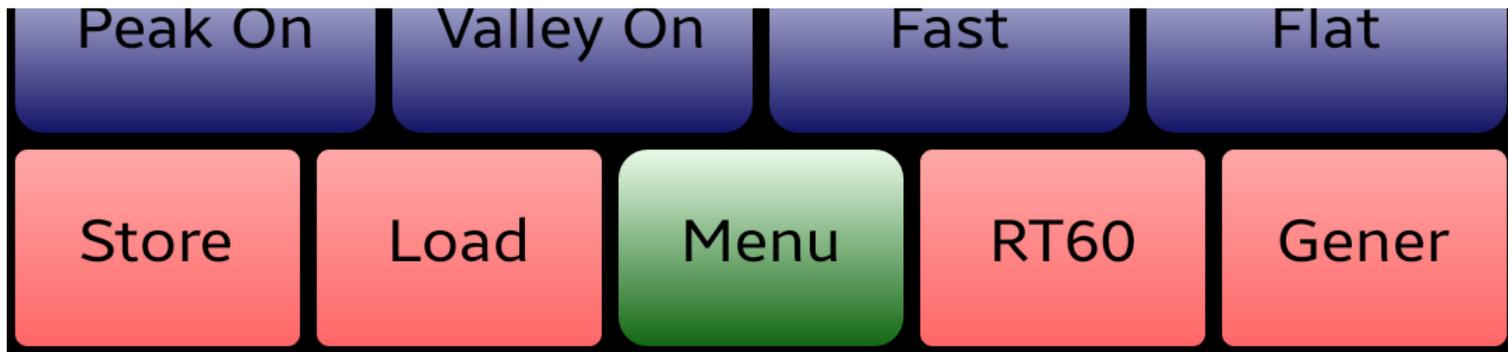
The real time analyzer (RTA) has several modes, described below.

### 1/1 Octave

Frequency response is shown in the ten ISO 1 octave bands.



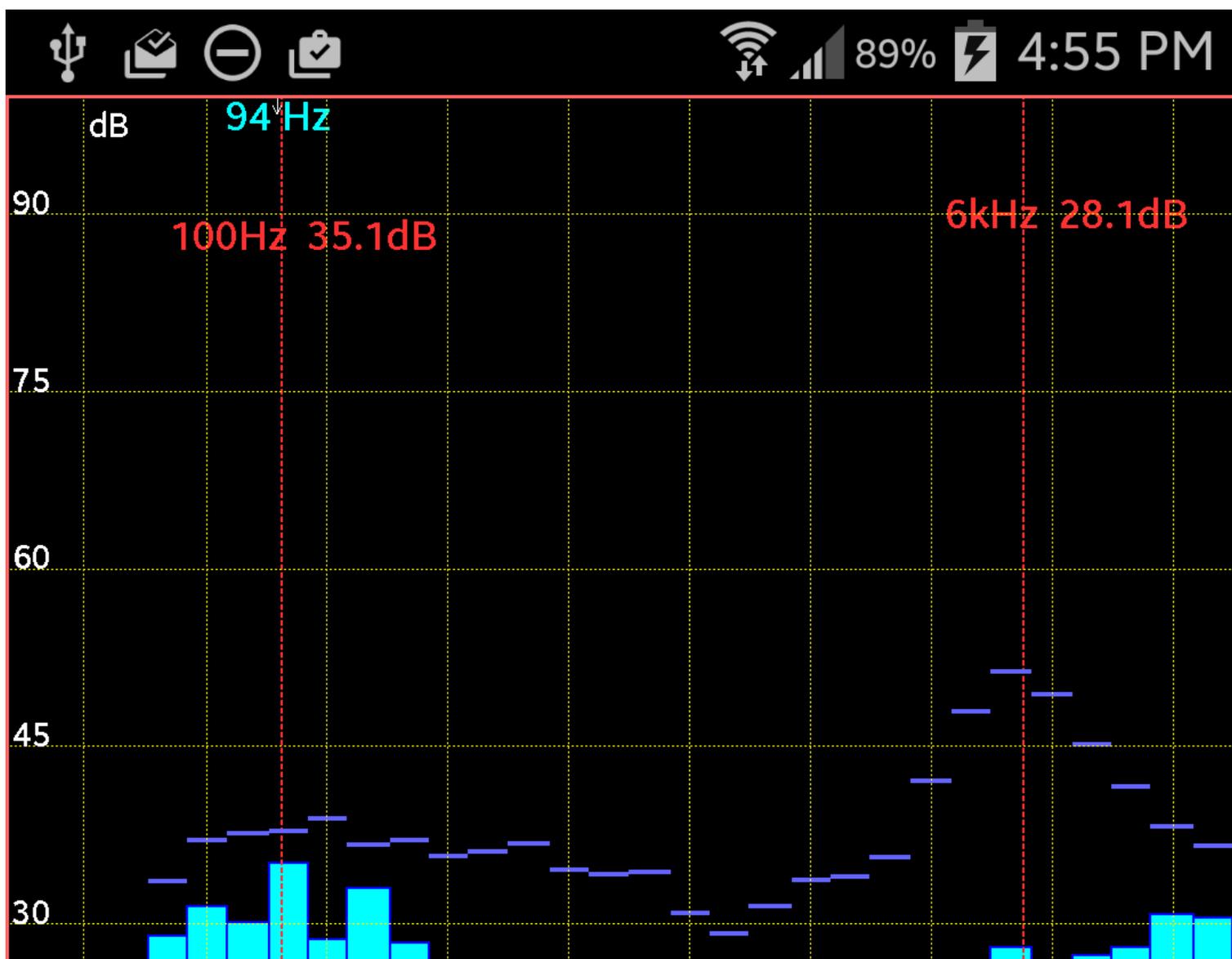


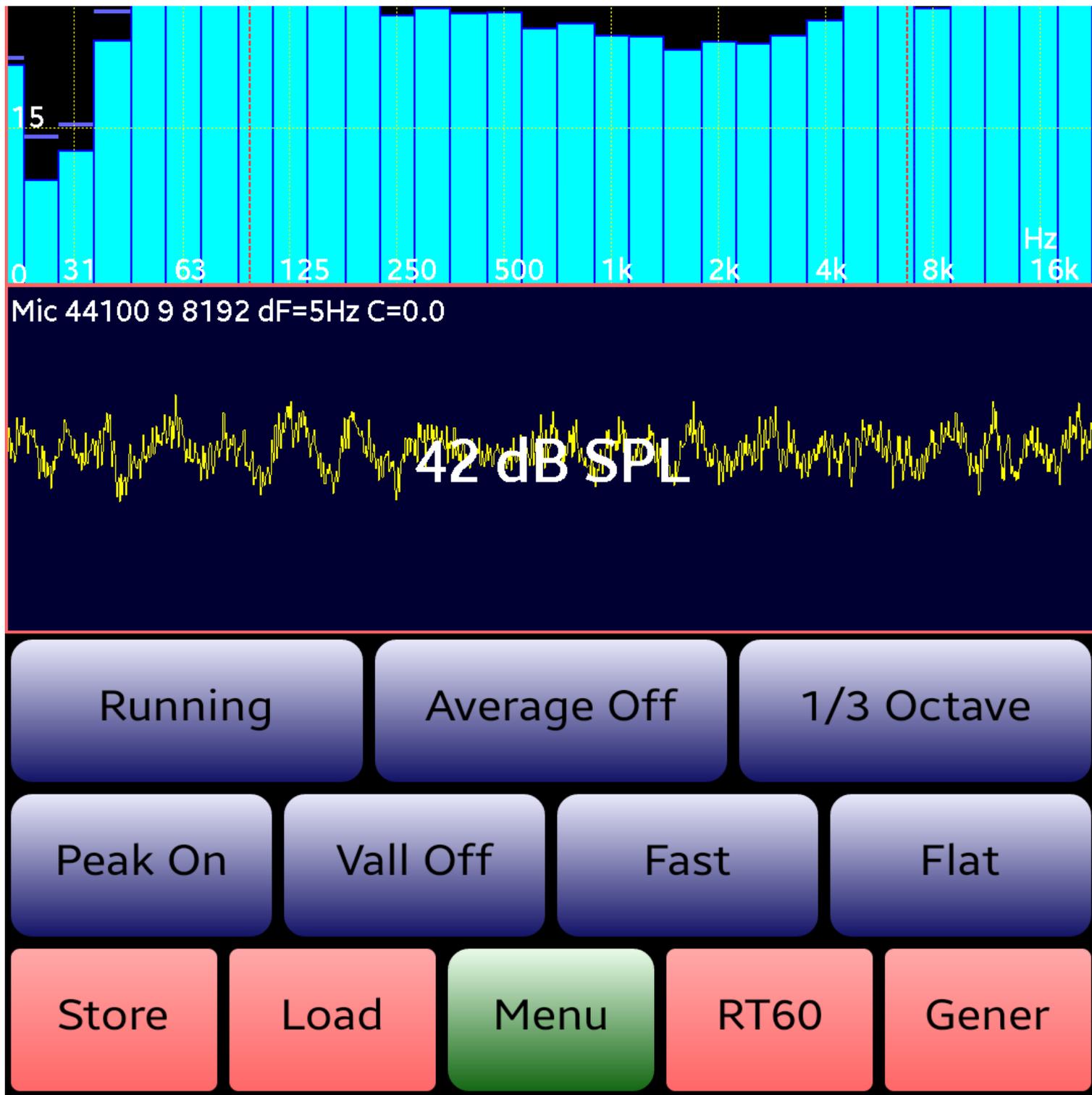


(In this screen shot, the Noise Curves are also being shown - see below for details on Noise Curves.)

### 1/3 Octave

Frequency response is shown in the thirty ISO 1/3 octave bands.

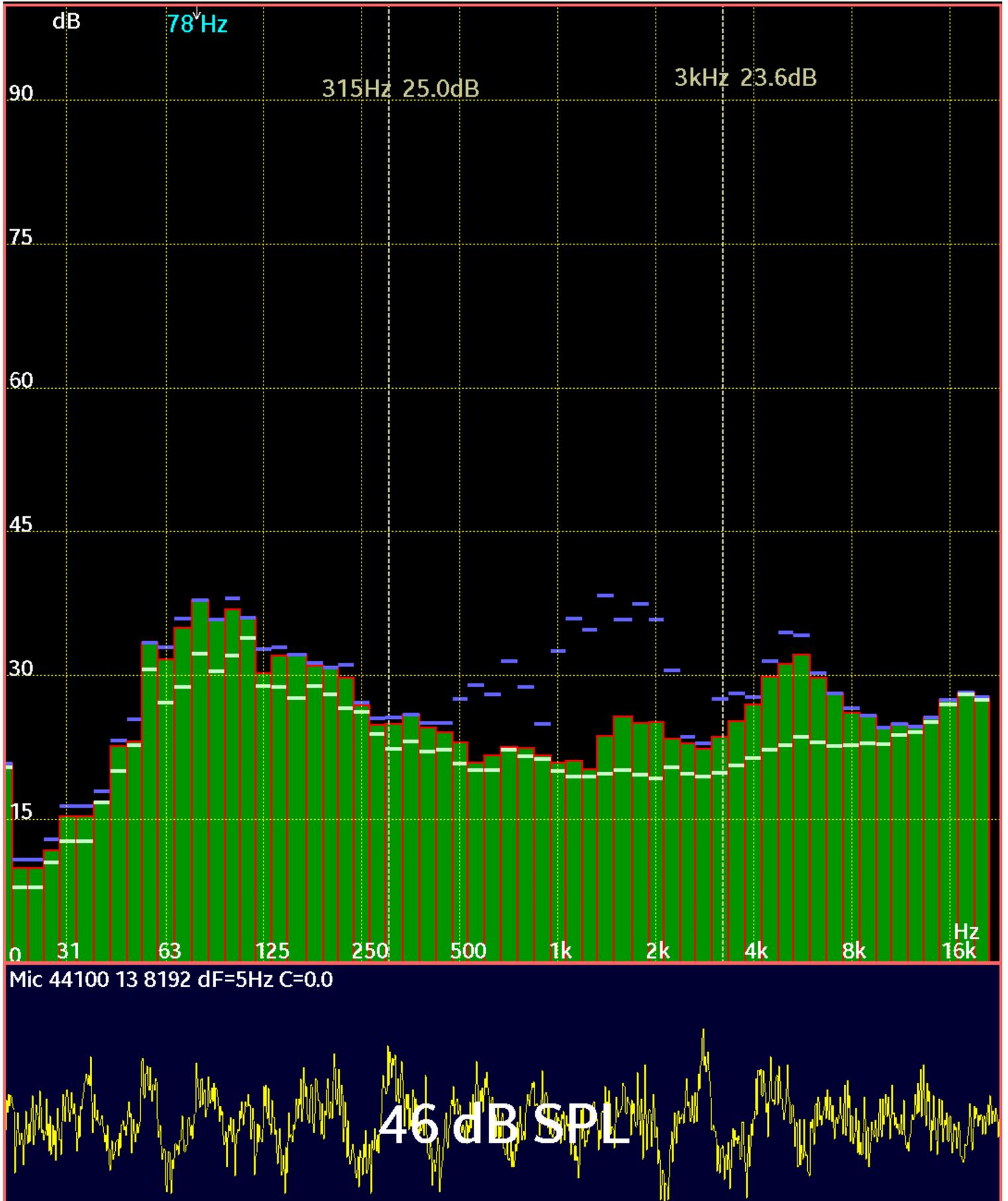


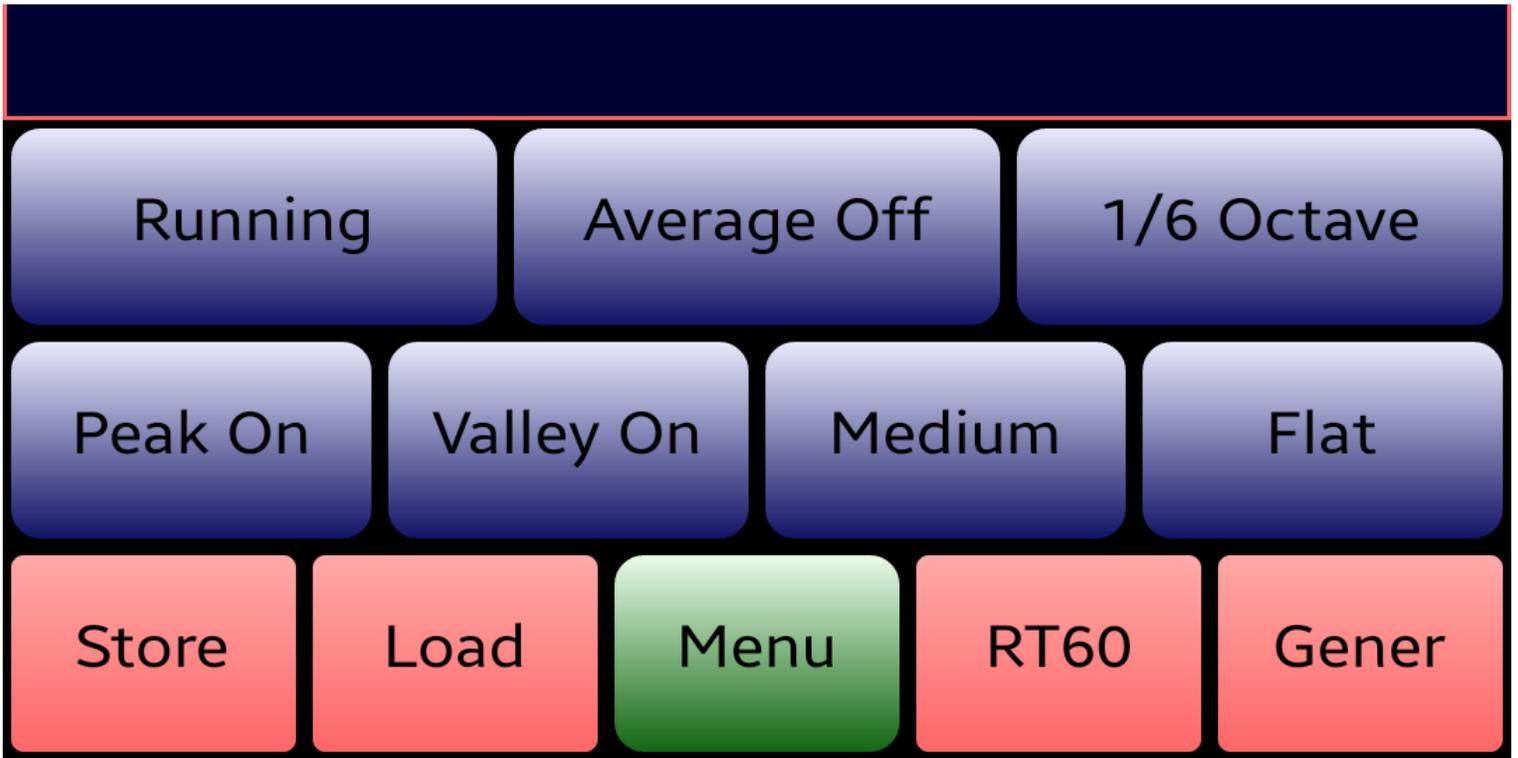


### 1/6 Octave

Frequency response is shown in ISO 1/6 octave bands.

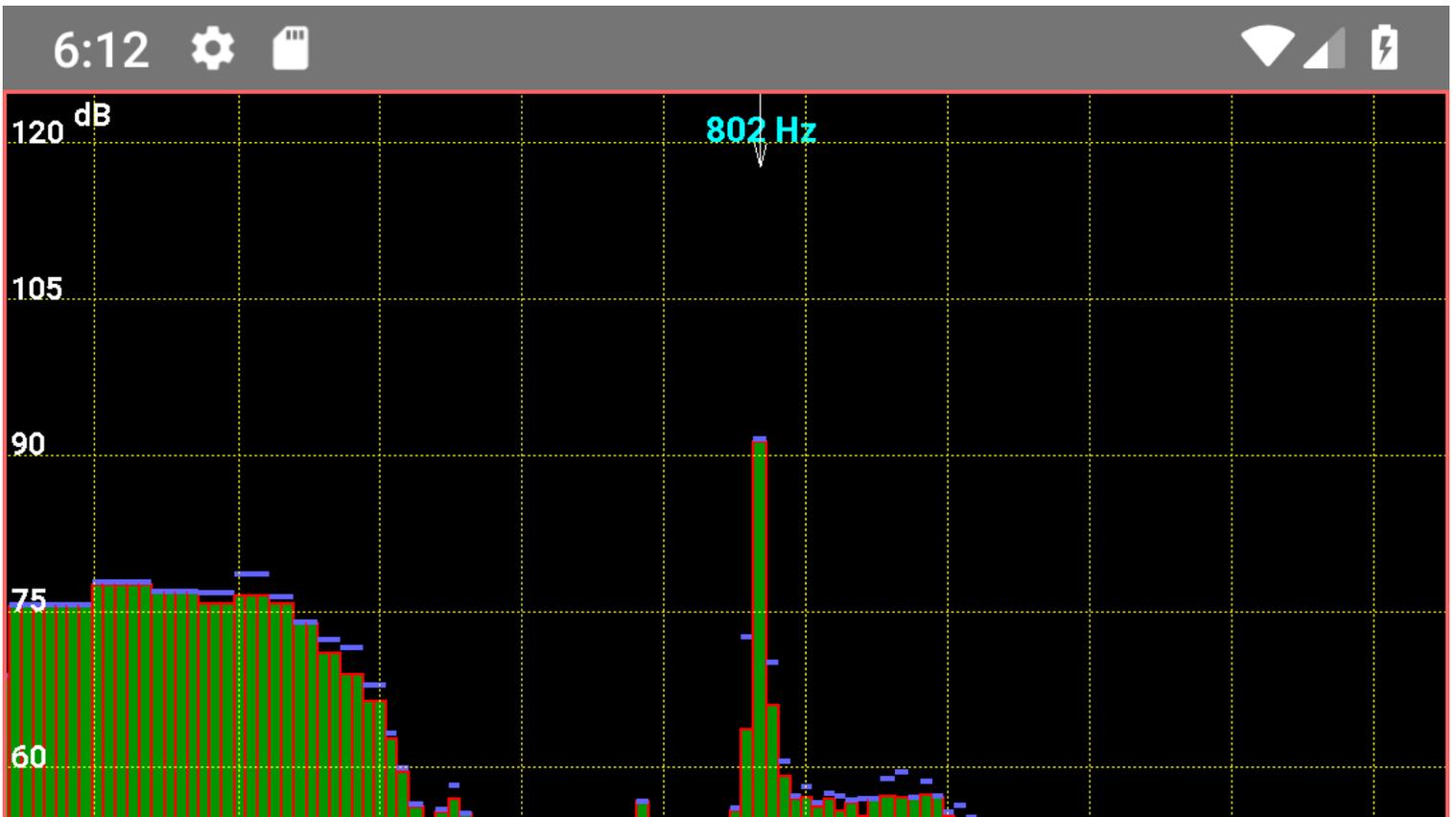


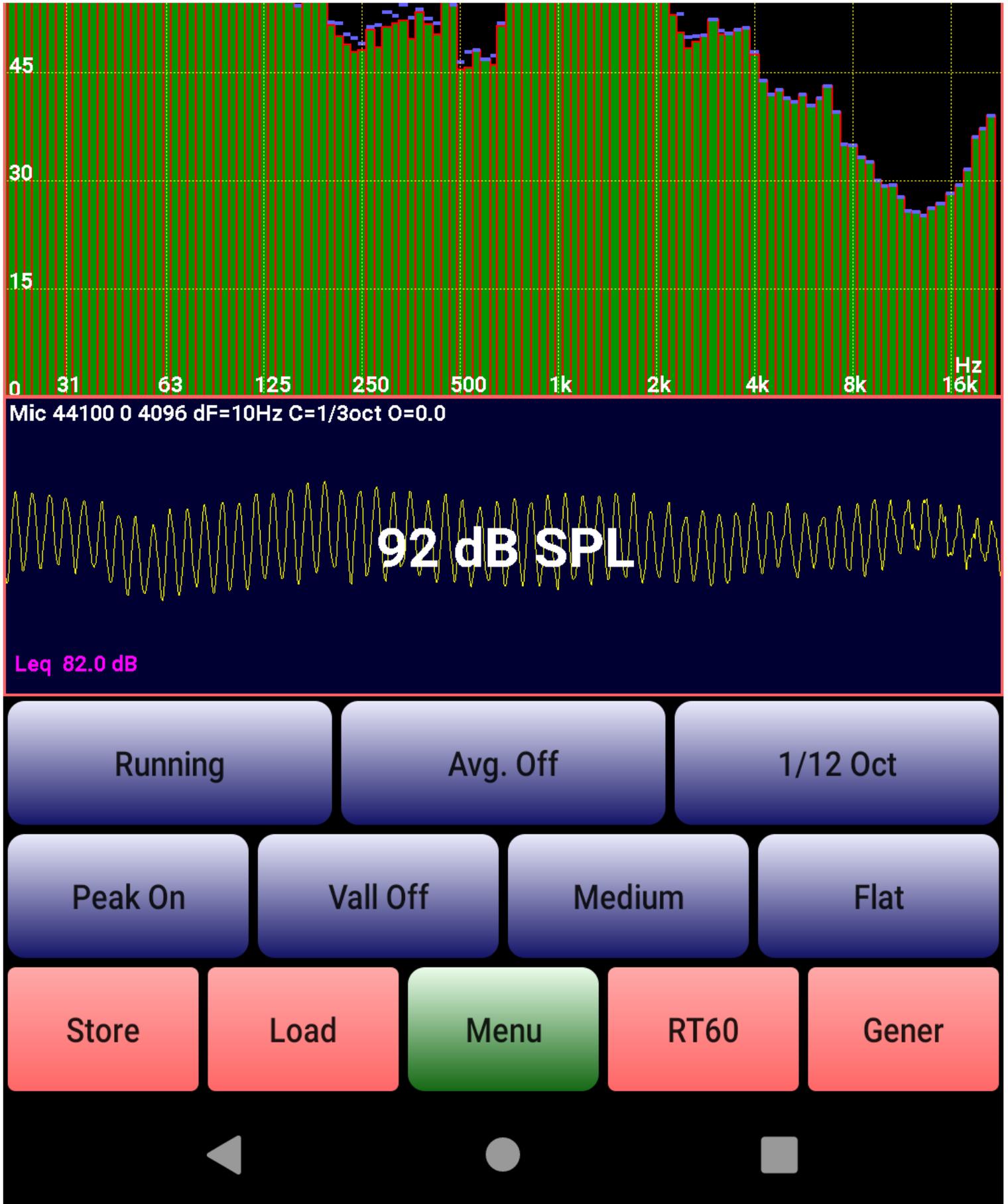




### 1/12 Octave

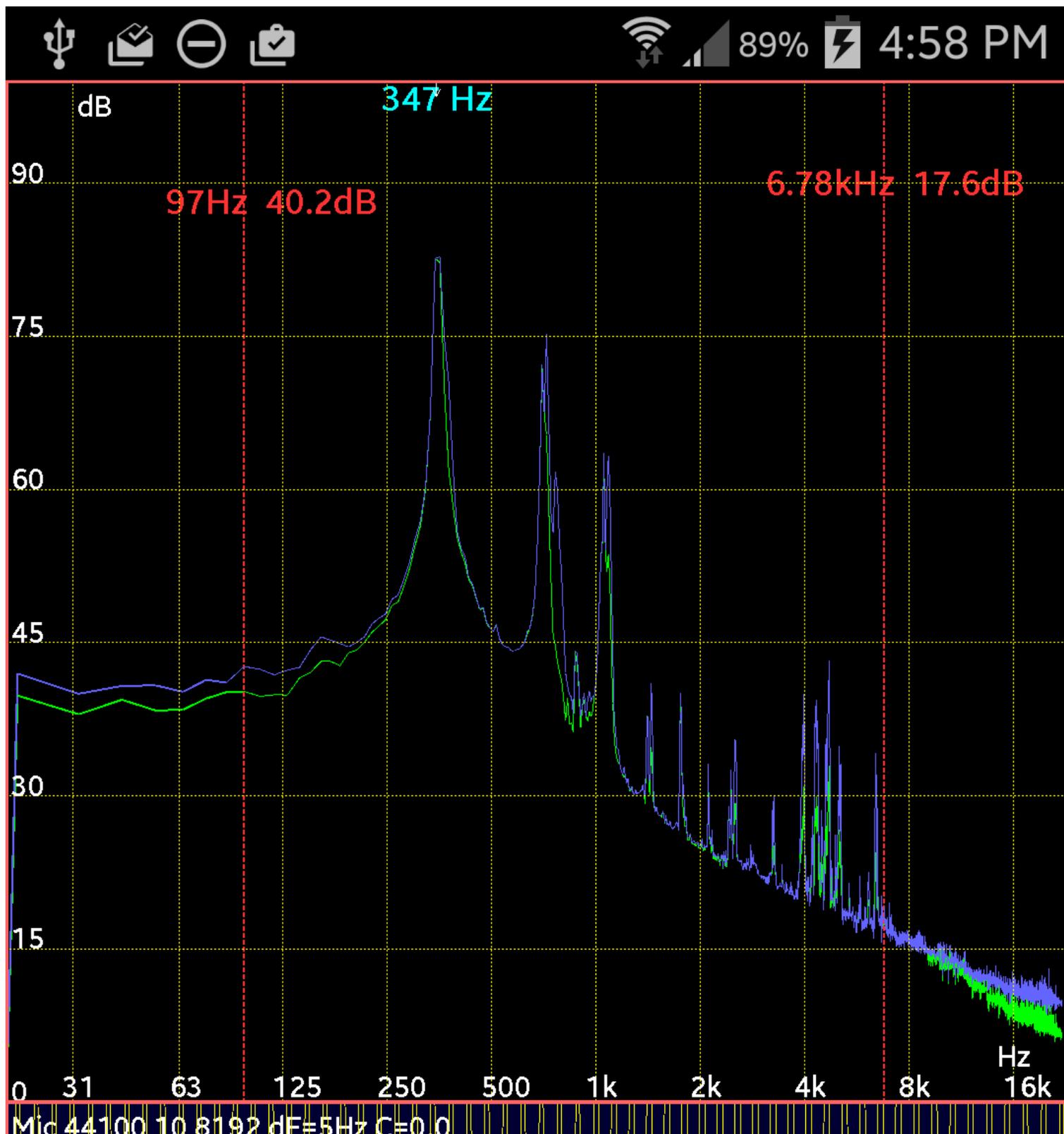
Frequency response is shown in ISO 1/12 octave bands.

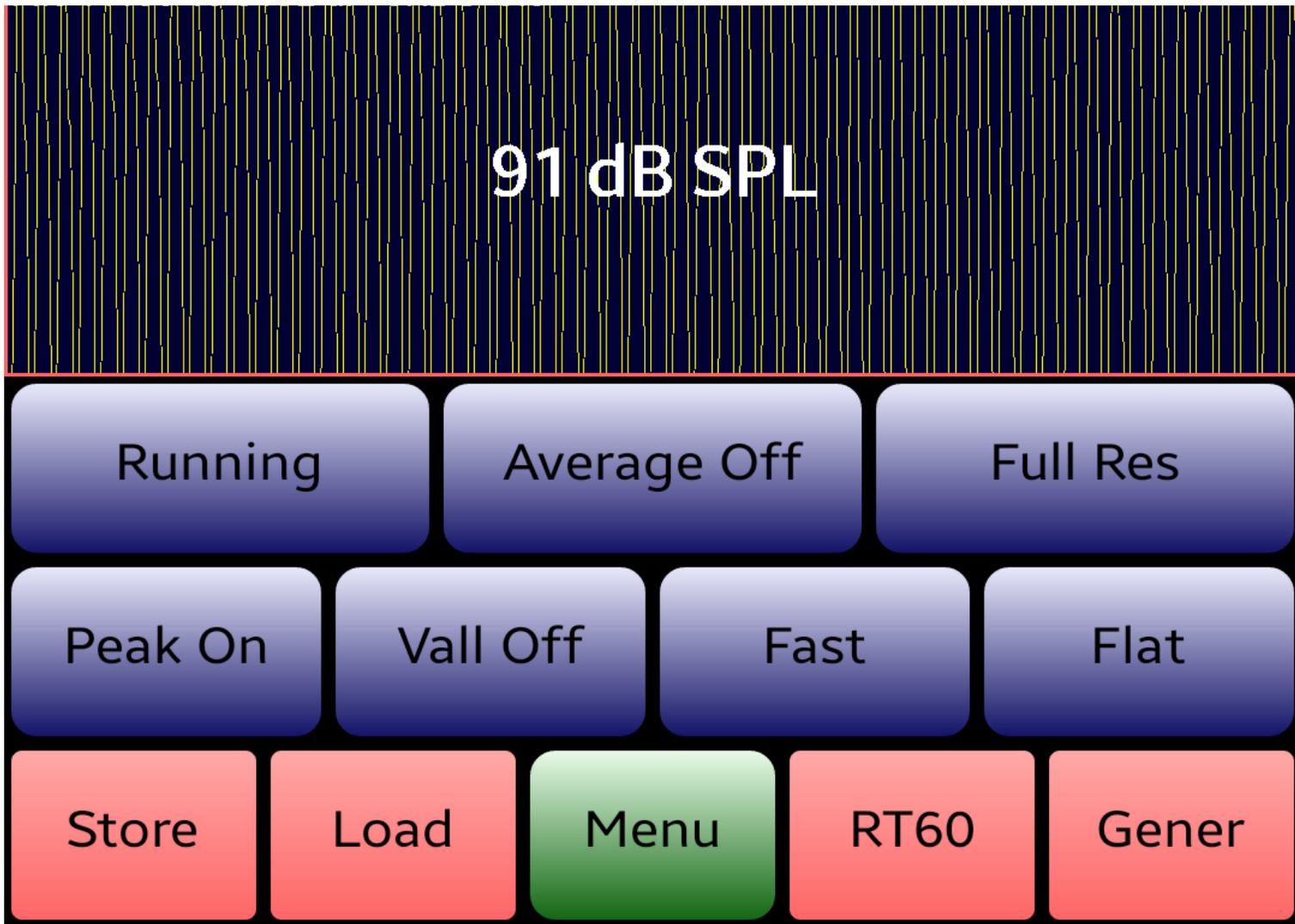




## Full Resolution

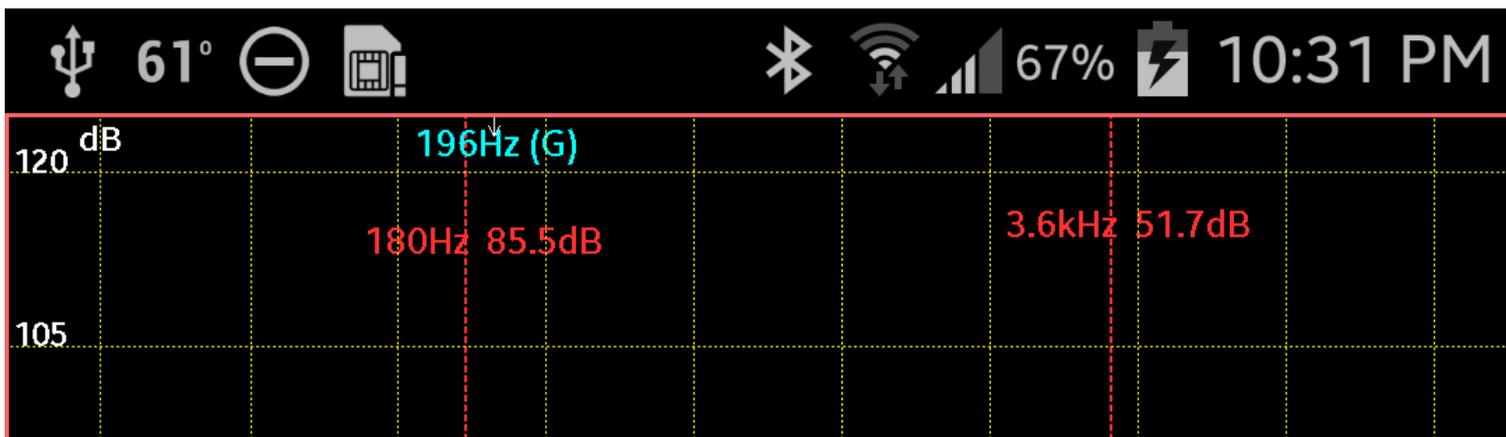
Frequency response is shown as a line graph at the maximum available resolution.

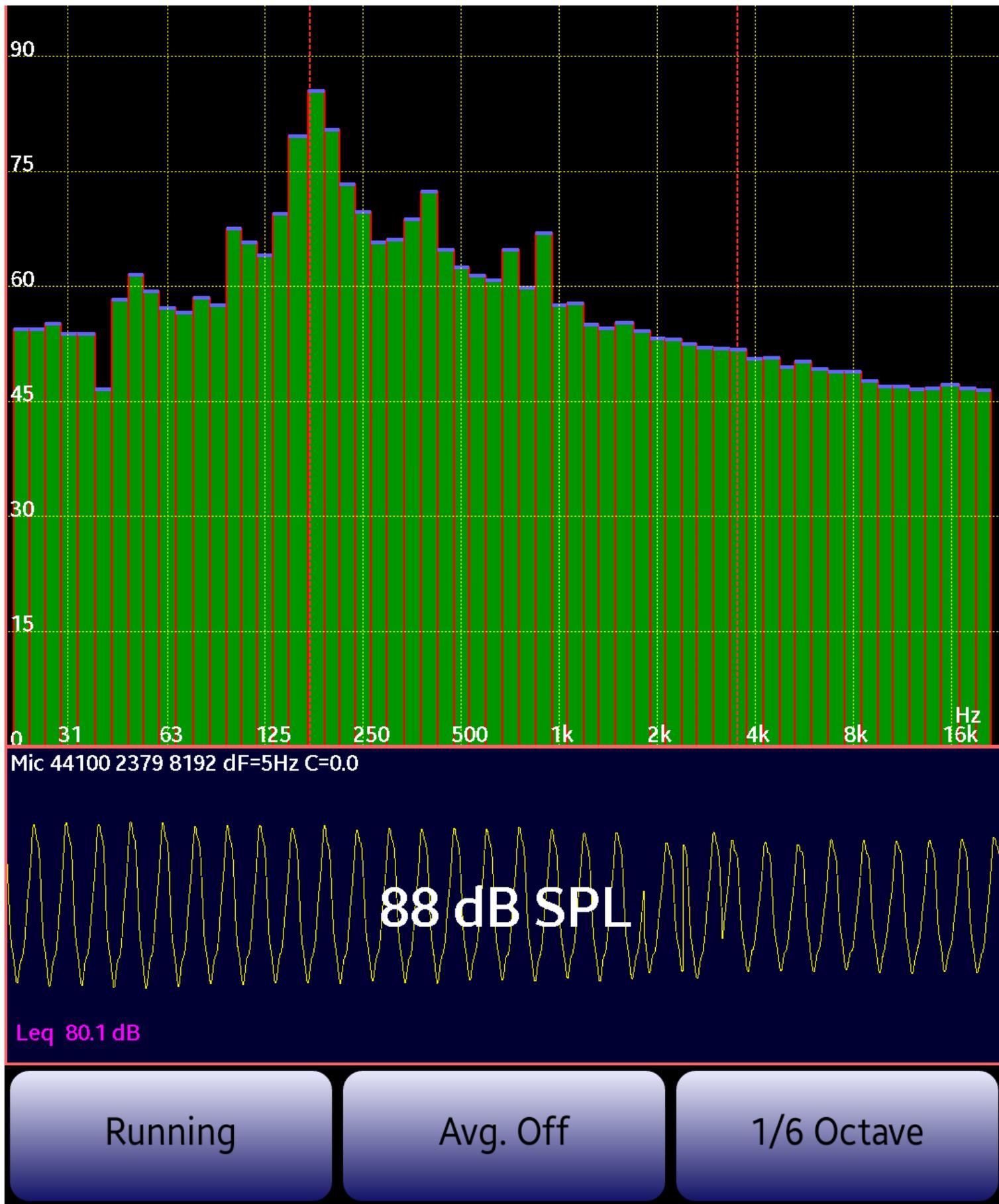


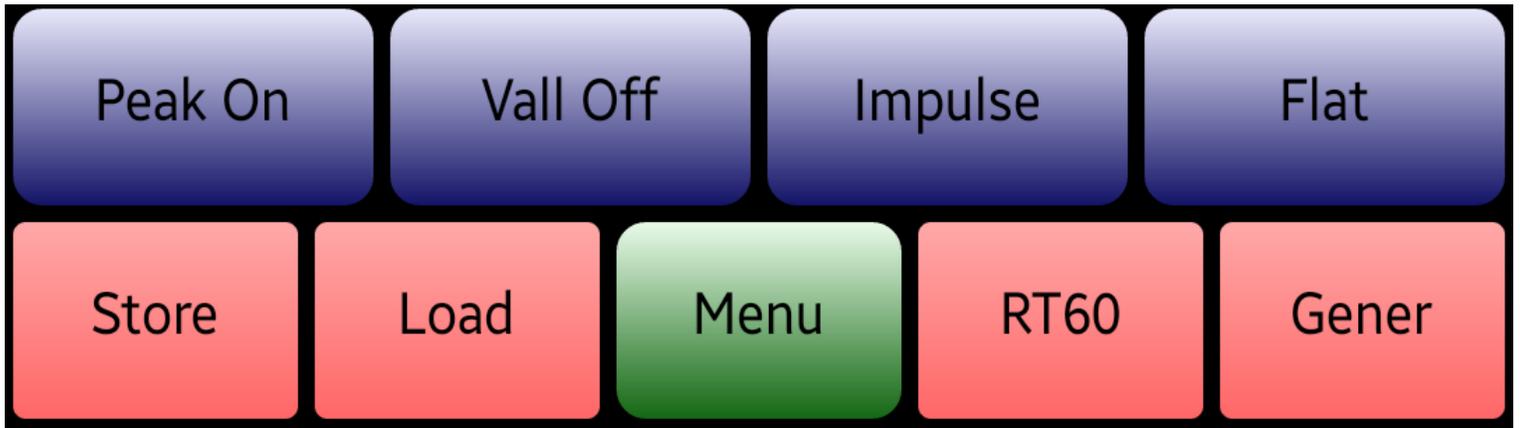


### Frequency of Max Intensity

In all the RTA modes, a marker at the top the screen shows the frequency with the maximum intensity. In the example below, it is showing 196Hz. The frequency is also shown as the note name on the musical scale - in this case "G". (The note name display can be turned off and on via the Menu.)

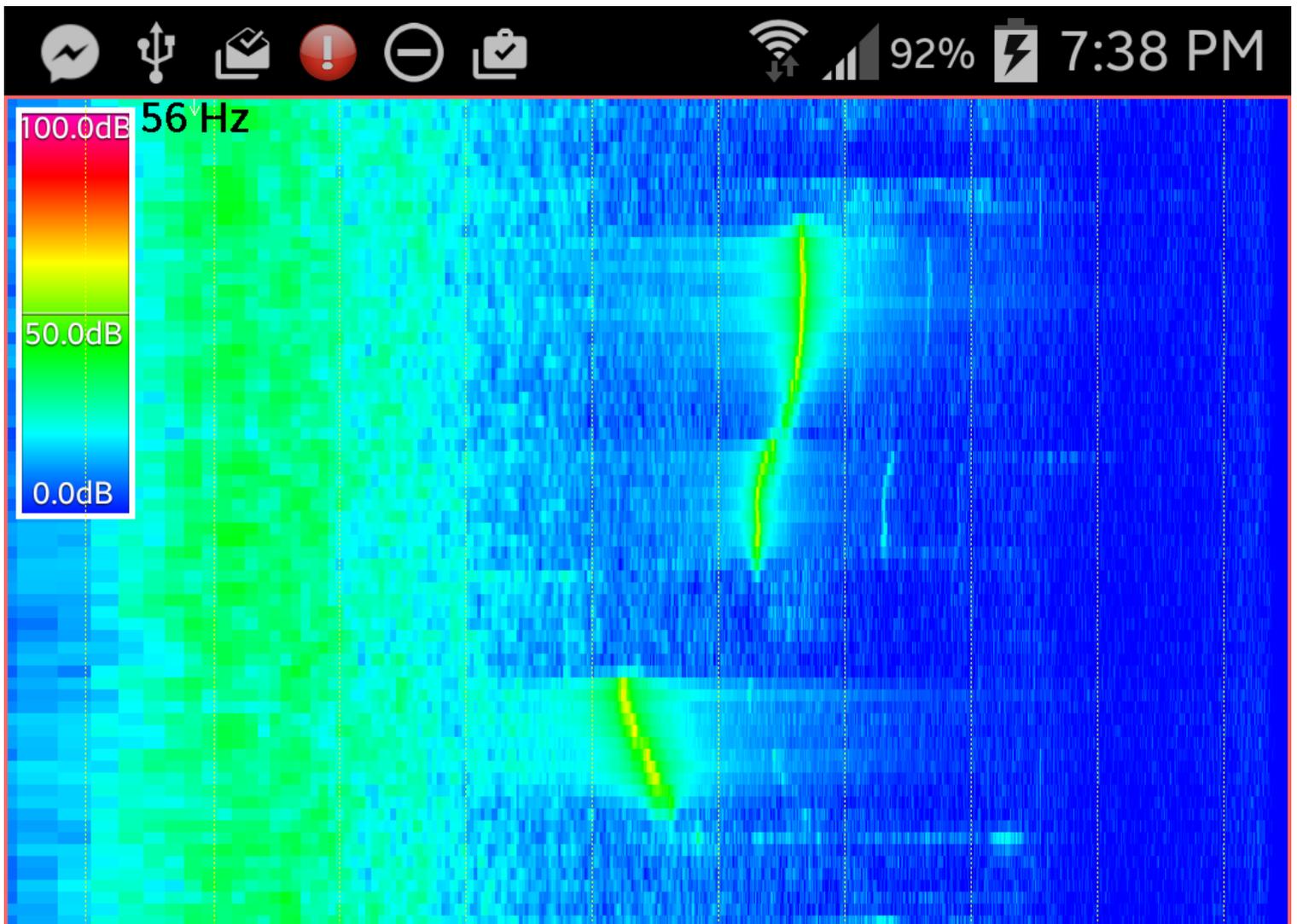


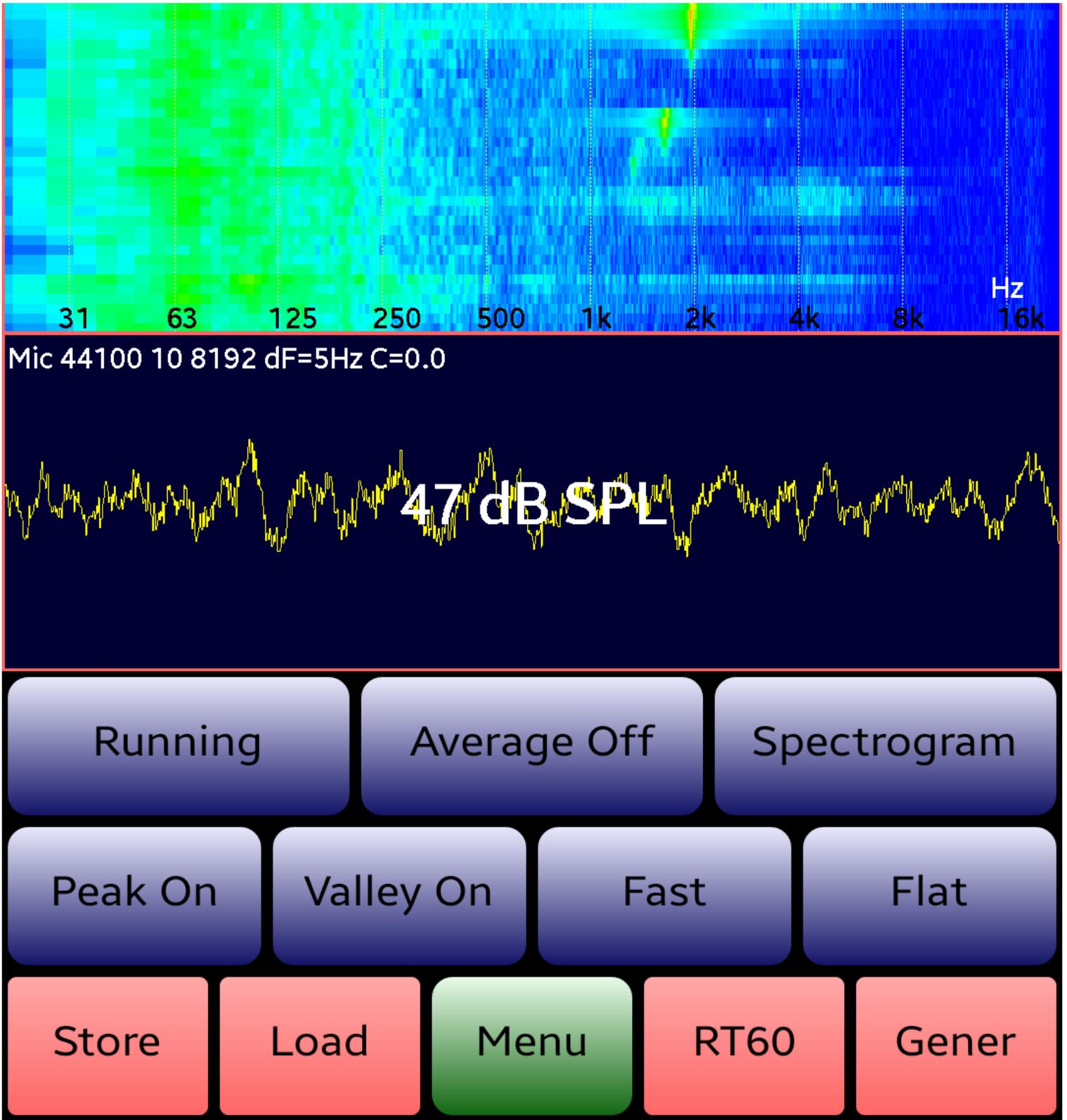




## Spectrogram

Frequency response is shown as a function of time: the spectrum at each moment is drawn as one row of a bitmap. The colour of each pixel indicates the intensity of the sound at that position in the spectrum. The most recent time is at the bottom of the display.

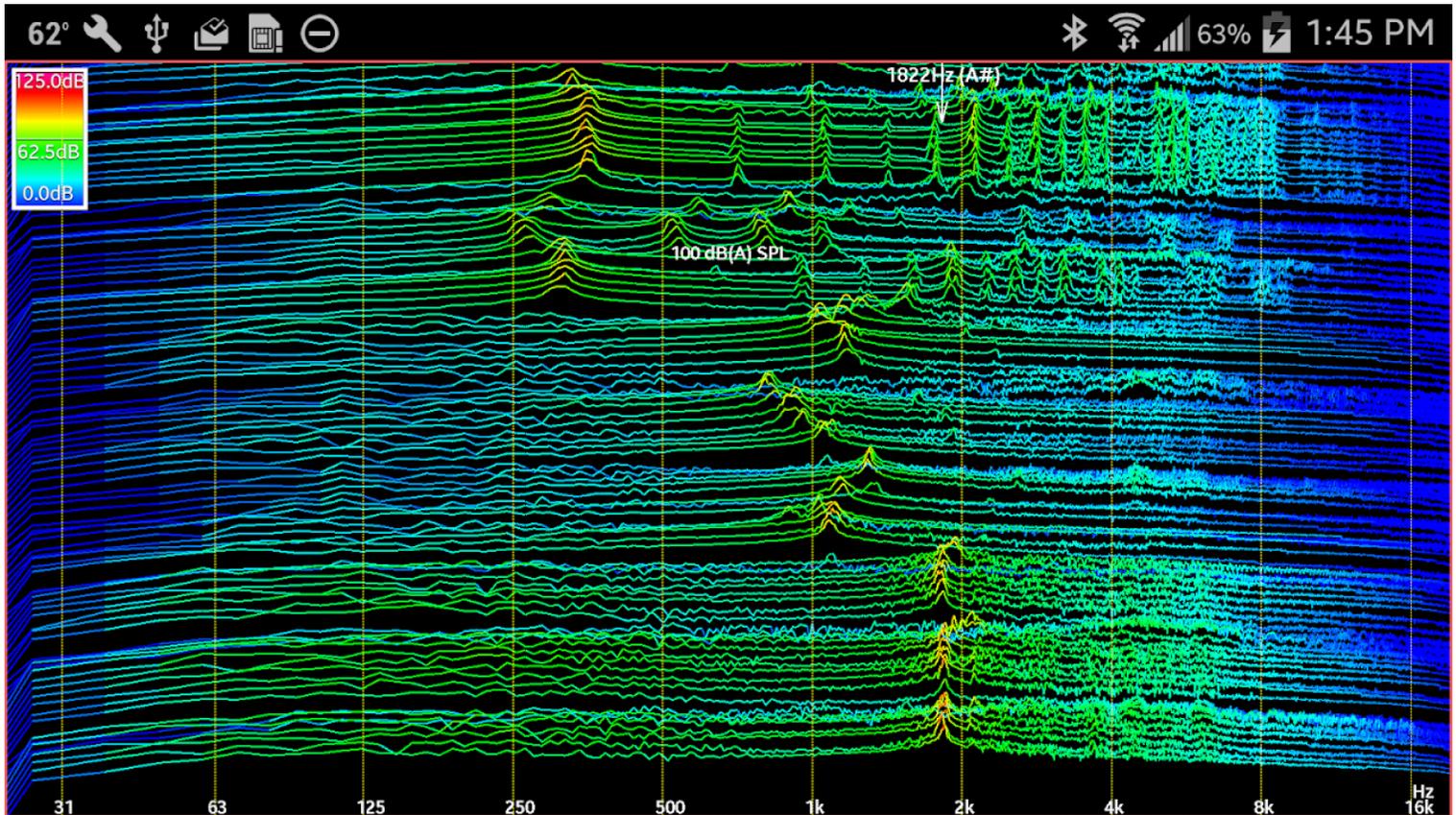




You can change the speed of the spectrogram (the rate at which new rows of the bitmap are drawn) between Fast and Slow using the option available in the Menu.

## Waterfall

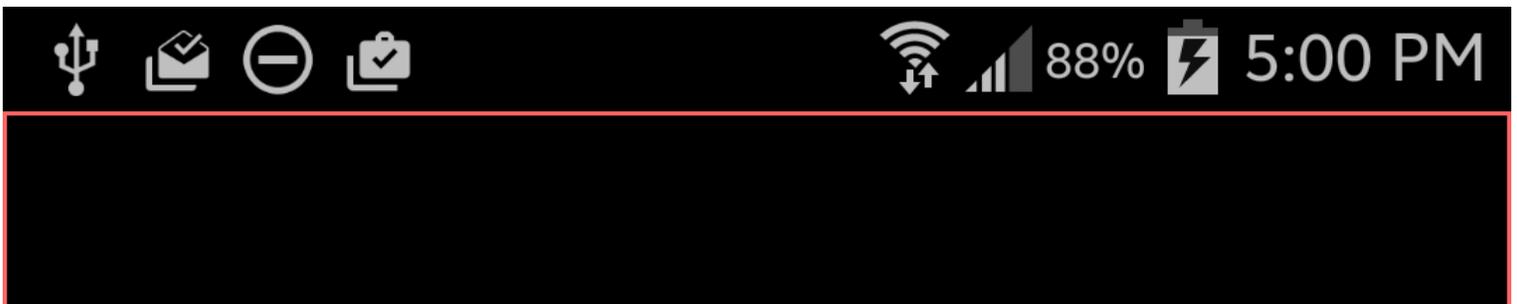
The waterfall display (introduced in v7.3) can be selected via the Menu, and replaces the Spectrogram display when selected.



The display builds up over time as new frequency response curves are added.

## SPL

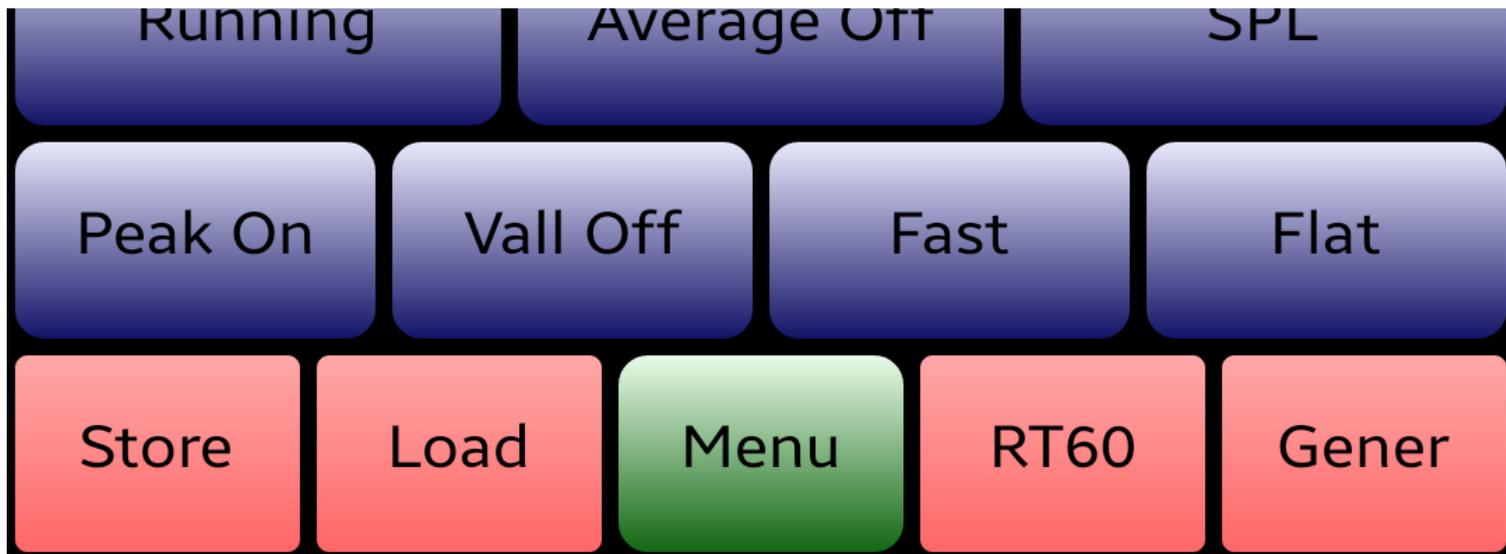
In this mode, the display simply shows the dB SPL figure.



46

Mic 44100 10 8192 dF=5Hz C=0.0





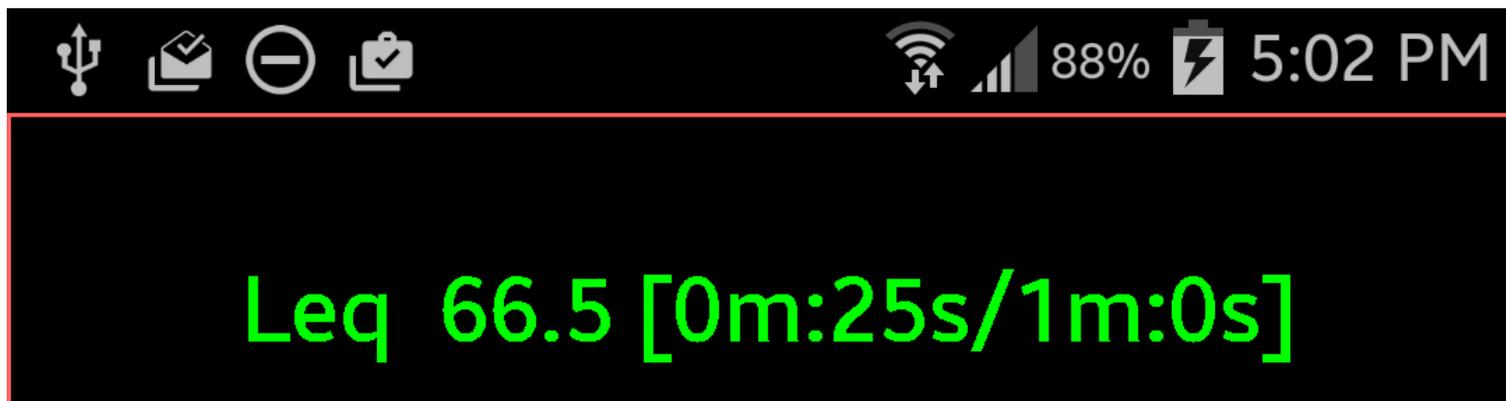
## Leq (Broadband and Octave)

In this mode, AudioTool shows the results of the continuous Leq calculation. Leq is the average of the power sum of the measured instantaneous dB SPL levels over an interval of time. For more details about Leq see, for example, [Leq at National Instruments](#). It is the "equivalent continuous sound level".

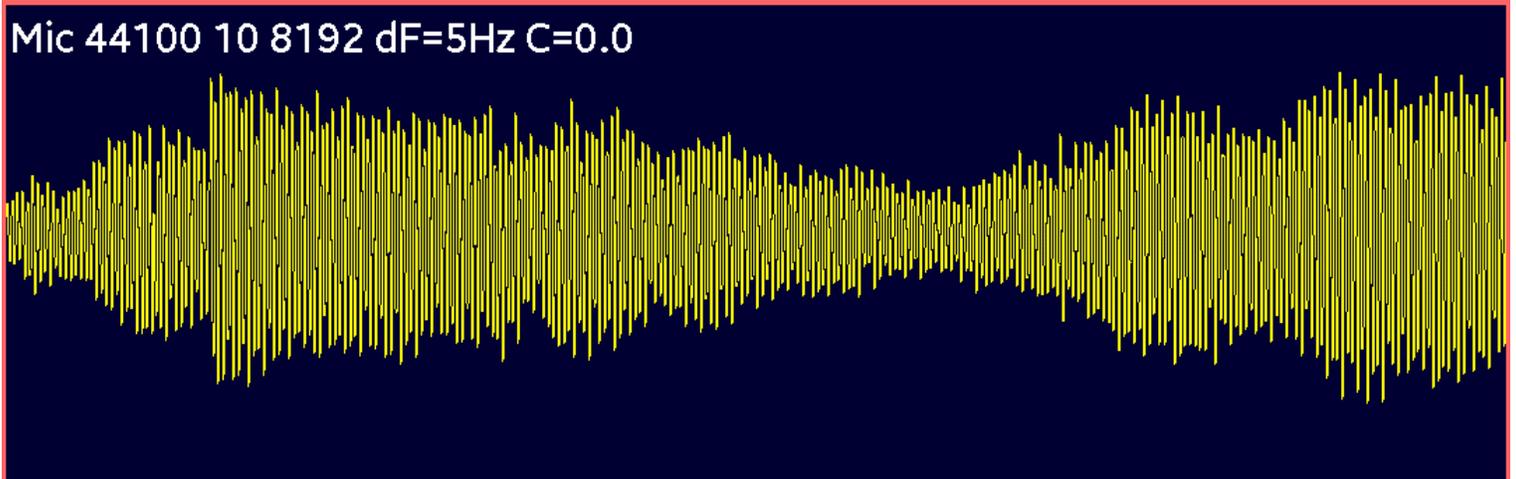
The Leq calculation will continue to run indefinitely, but it can be stopped by pressing the RTA "Running" button (top left) - if you press that button again, a new Leq calculation is started afresh.

Shown below is a screenshot of the Leq display. The broadband Leq, and the elapsed time for the calculation is shown, as well as the target duration (in this case 1 minute). (The duration can be adjusted using the Menu option "Set Leq Duration".)

The table shows the 1 octave Leq measurements: the current calculation is shown in the "Now" column. The other columns show prior results ... so the T-1 column is the result from the calculation that last finished, the T-2 the one before that, and so on.



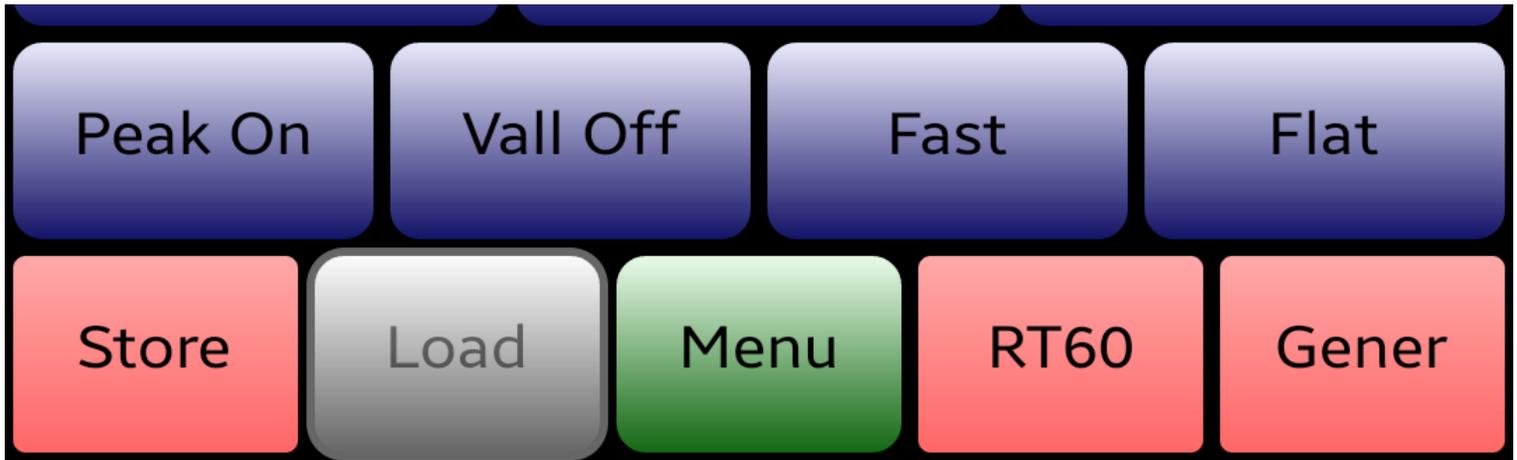
	Now	T-1	T-2	T-3	T-4	T-5
<b>Broad</b>	66.5	56.8				
<b>31Hz</b>	41.6	38.0				
<b>63Hz</b>	54.1	47.1				
<b>125Hz</b>	53.6	45.9				
<b>250Hz</b>	53.5	46.2				
<b>500Hz</b>	47.9	47.6				
<b>1kHz</b>	65.1	43.2				
<b>2kHz</b>	50.4	41.6				
<b>4kHz</b>	50.2	49.2				
<b>8kHz</b>	52.0	51.8				
<b>16kHz</b>	40.4	40.5				



Running

Average Off

Leq



Note that, if you change filters, then the Leq integration re-starts from the moment the filter was changed.

(Normally, Leq measurements are made using the A Weight filter.)

### Storing Leq Measurements

AudioTool stores up to 5,000 Leq measurements internally starting from every time the app is started. The measurements can be saved to an AudioTool .leq file - use the "Store" button for this purpose. The stored file can be opened in a text editor, and imported into e.g. Excel. Here is an extract from an Leq file:

AudioTool Leq File recorded at Sat Jun 15 11:42:18 PDT 2013

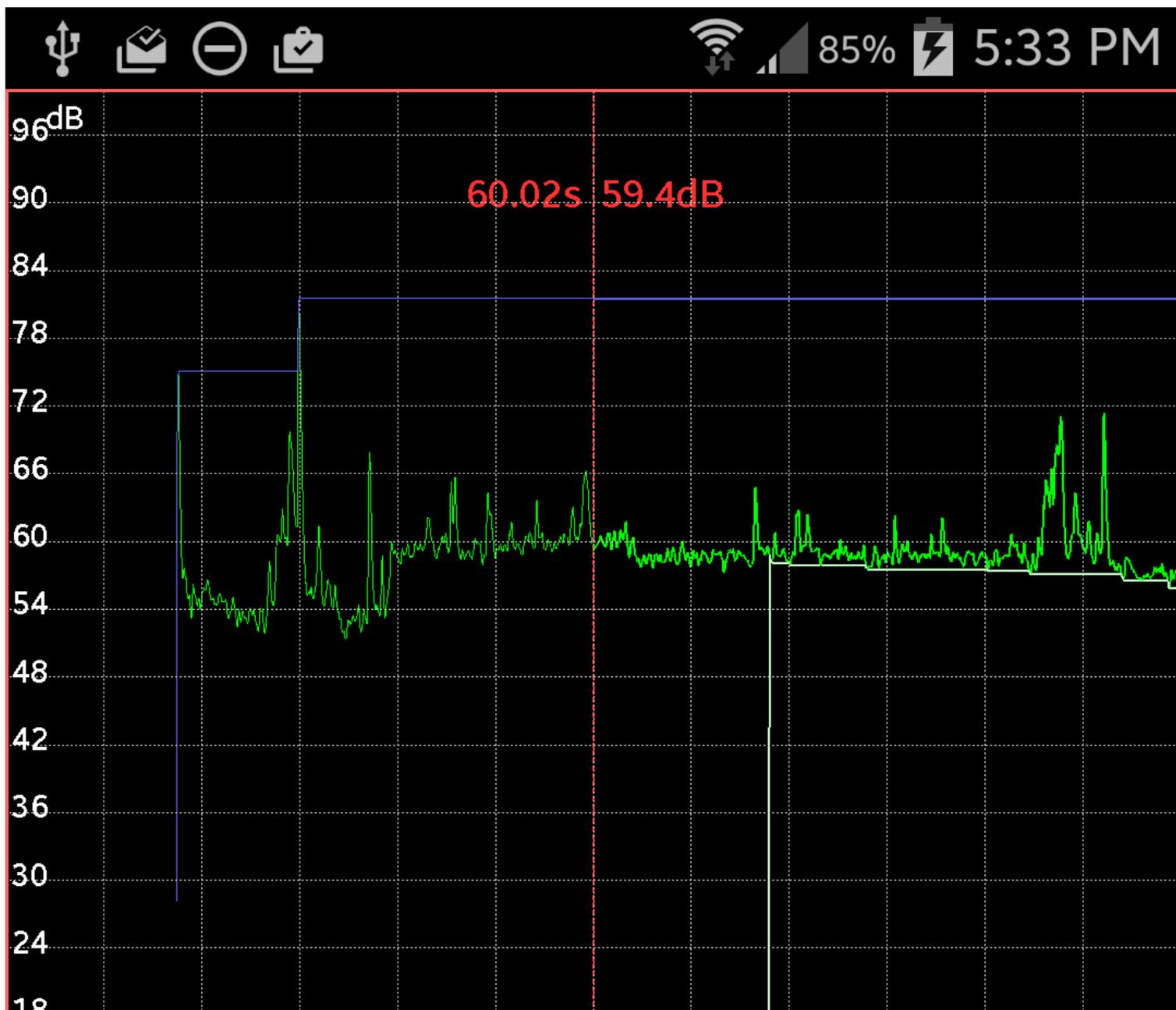
MilliSeconds now: 1371321738067 Columns: Time Start, Time Finish, Duration(s), LEQ, LEQ octave1, LEQ octave2 ... 40 LEQ measurements in file

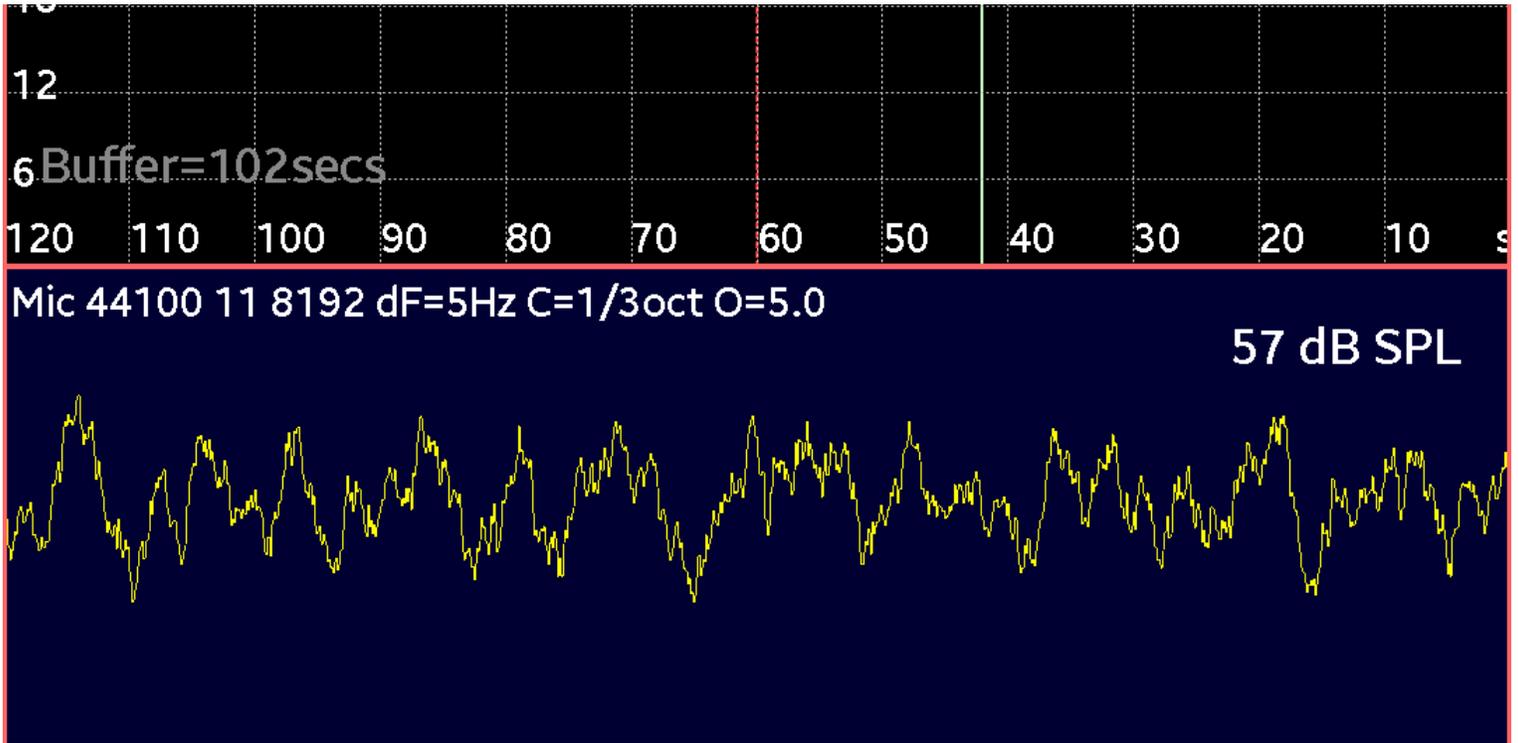
```
2013-06-15T11:41:13 2013-06-15T11:42:13 60 74.4 47.4 50.9 60.7 65.8 66.6 65.1 64.3 65.9 68.8 58.1
2013-06-15T11:40:13 2013-06-15T11:41:13 60 69.0 43.9 47.0 58.9 63.7 61.8 59.3 53.2 54.1 62.5 51.2
2013-06-15T11:39:13 2013-06-15T11:40:13 60 65.8 43.3 46.4 57.8 60.0 58.7 55.8 49.5 50.4 58.4 51.2
2013-06-15T11:38:13 2013-06-15T11:39:13 60 66.1 43.1 47.0 57.7 62.3 58.4 55.7 49.4 49.2 55.0 51.1
2013-06-15T11:37:13 2013-06-15T11:38:13 60 69.3 44.0 53.5 58.1 63.3 61.0 59.0 54.2 55.6 64.2 52.1
2013-06-15T11:36:13 2013-06-15T11:37:13 60 69.1 43.2 50.3 58.8 63.5 59.4 57.5 51.1 52.6 64.8 52.2
2013-06-15T11:35:13 2013-06-15T11:36:13 60 67.9 44.9 50.2 56.8 62.3 59.2 59.1 53.2 54.7 61.9 51.8
```

2013-06-15T11:34:13 2013-06-15T11:35:13 60 69.0 45.9 51.0 58.9 64.1 60.1 60.5 55.3 54.9 61.4 52.7

## Chart Recorder

The time history of SPL readings is shown: the chart recorder "pen" at the right hand side of the screen draws the latest value. The axis is labelled in seconds in the past. At most 10,000 SPL readings are stored, corresponding to a few minutes of history (this value varies from one Android device to another). The chart data can be stored using the Store button, if so desired. (Note the use of the cursors in the example below.) The grey text at the bottom left of the chart indicates the current length of the buffer, in seconds.





Running	Average Off	Chart		
Peak On	Vall On	Fast	Flat	
Store	Load	Menu	RT60	Gener

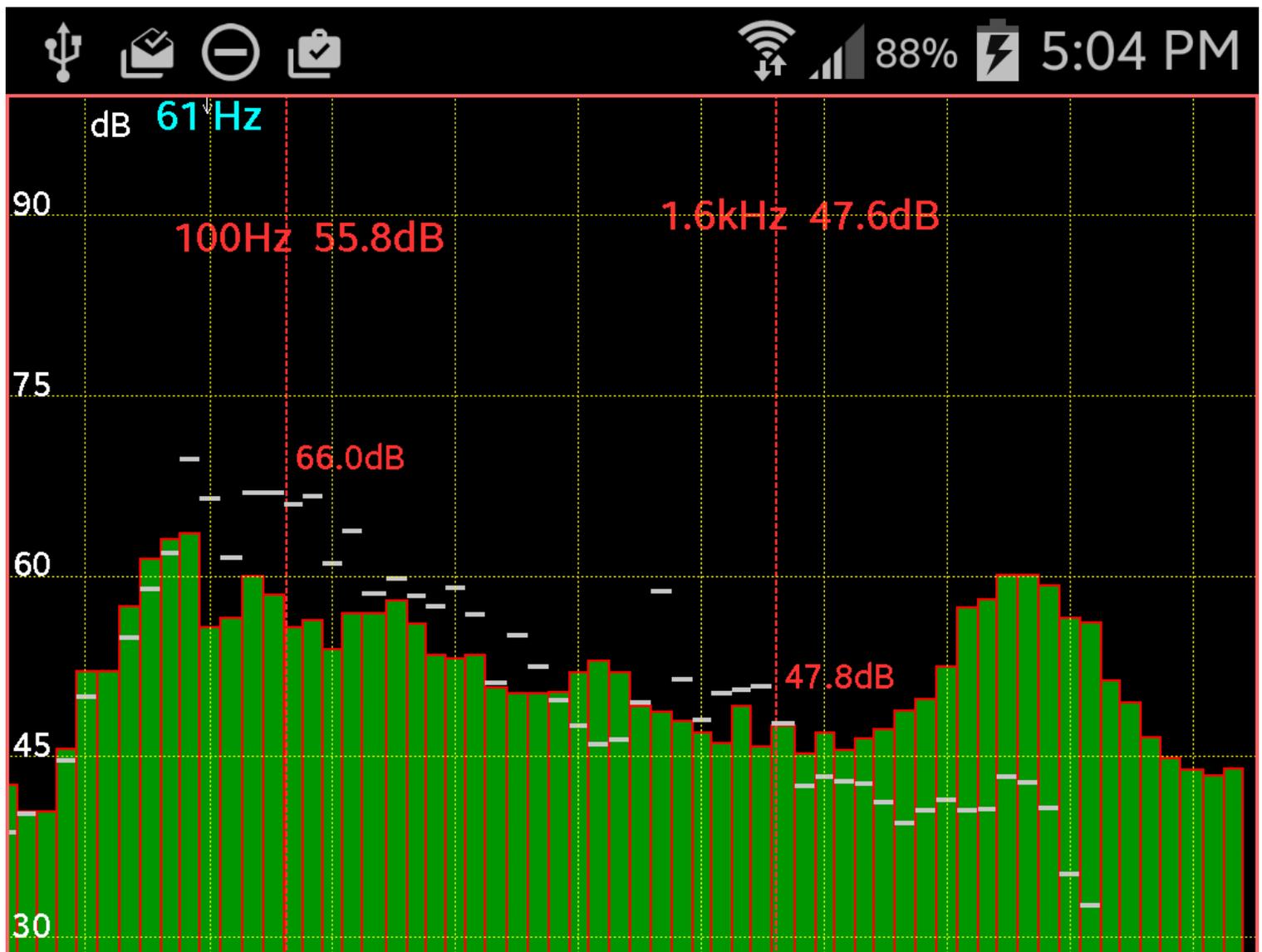
## Peak Store (for Sweeps)

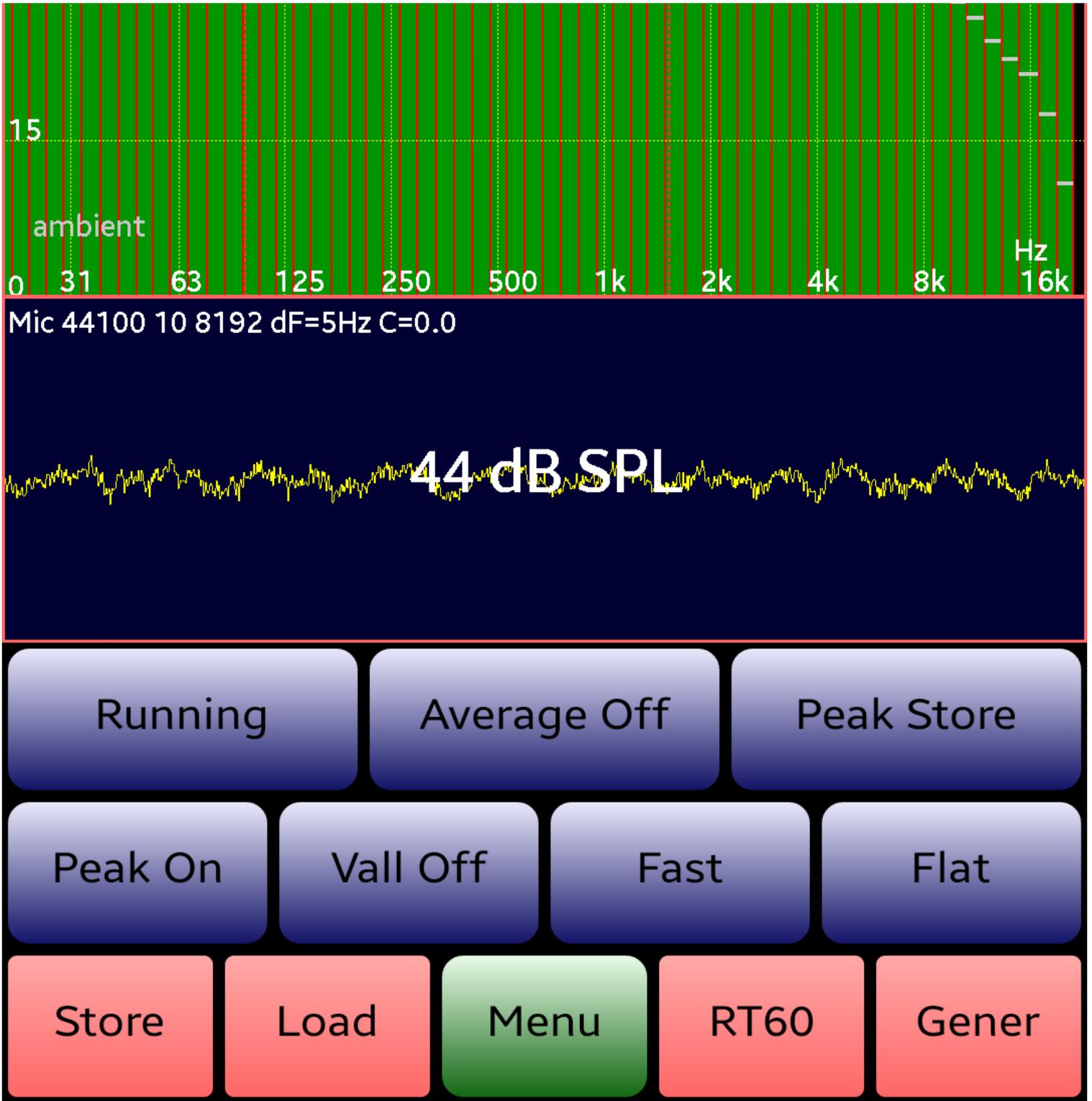
In this mode, AudioTool keeps track of the peak dB values at all measured frequencies. This is useful for e.g. sweeping the frequency response of an audio system. Peak Store mode is a 1/6 octave mode.

To make such a measurement, set the RTA mode to Peak Store (see the screenshot below), and then go to the Generator, select either Sweep or Log Sweep, configure the frequency range you desire, then start the Generator running.

Return to the Analyzer, and watch the peaks set as the Generator sweeps across the frequencies you specified. In this way, the frequency response of the system may be measured.

To reset the peak dB values, stop the data collection by pressing the "Running" button, and press it again to set the peaks to zero and start a new measurement.

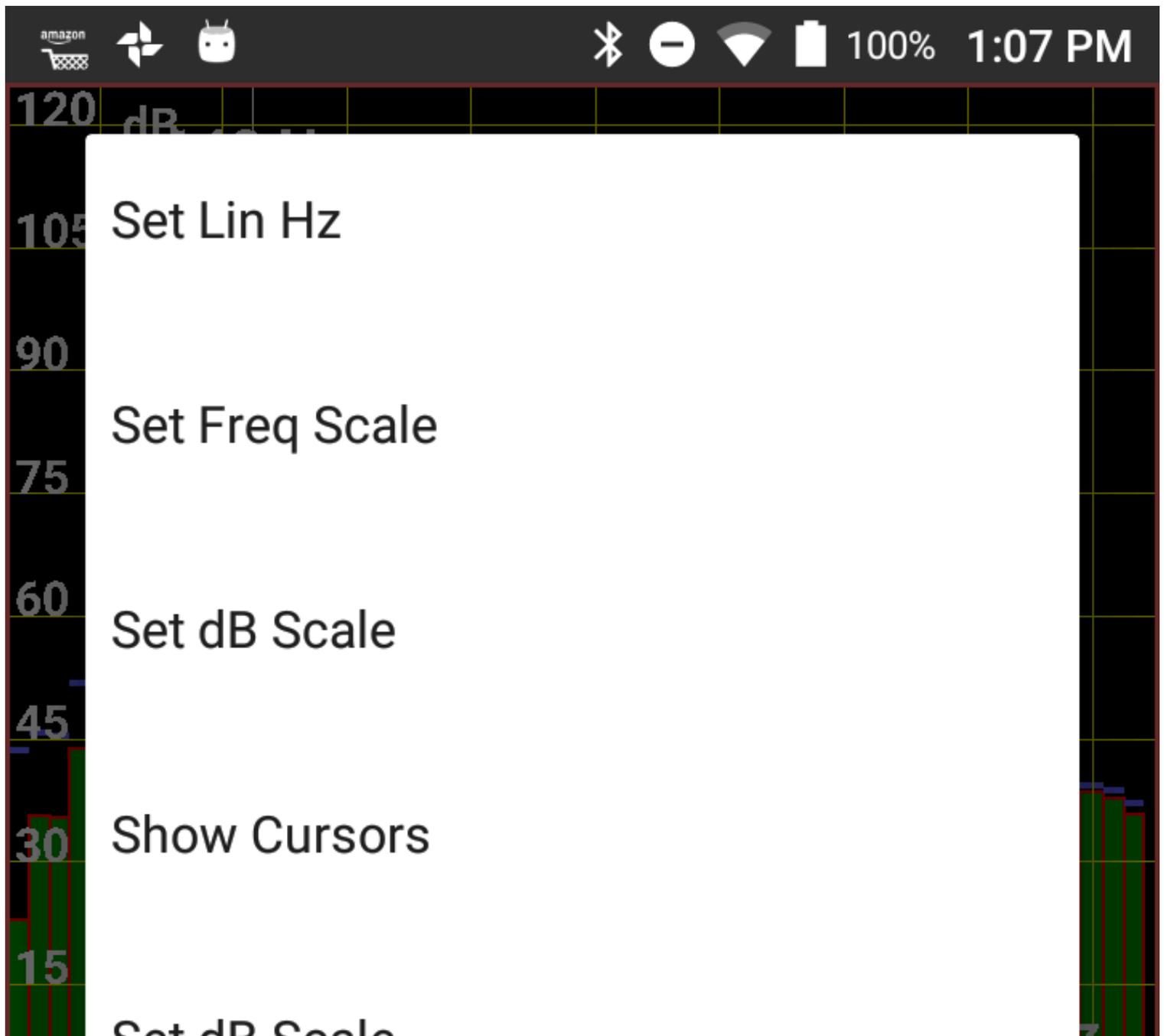


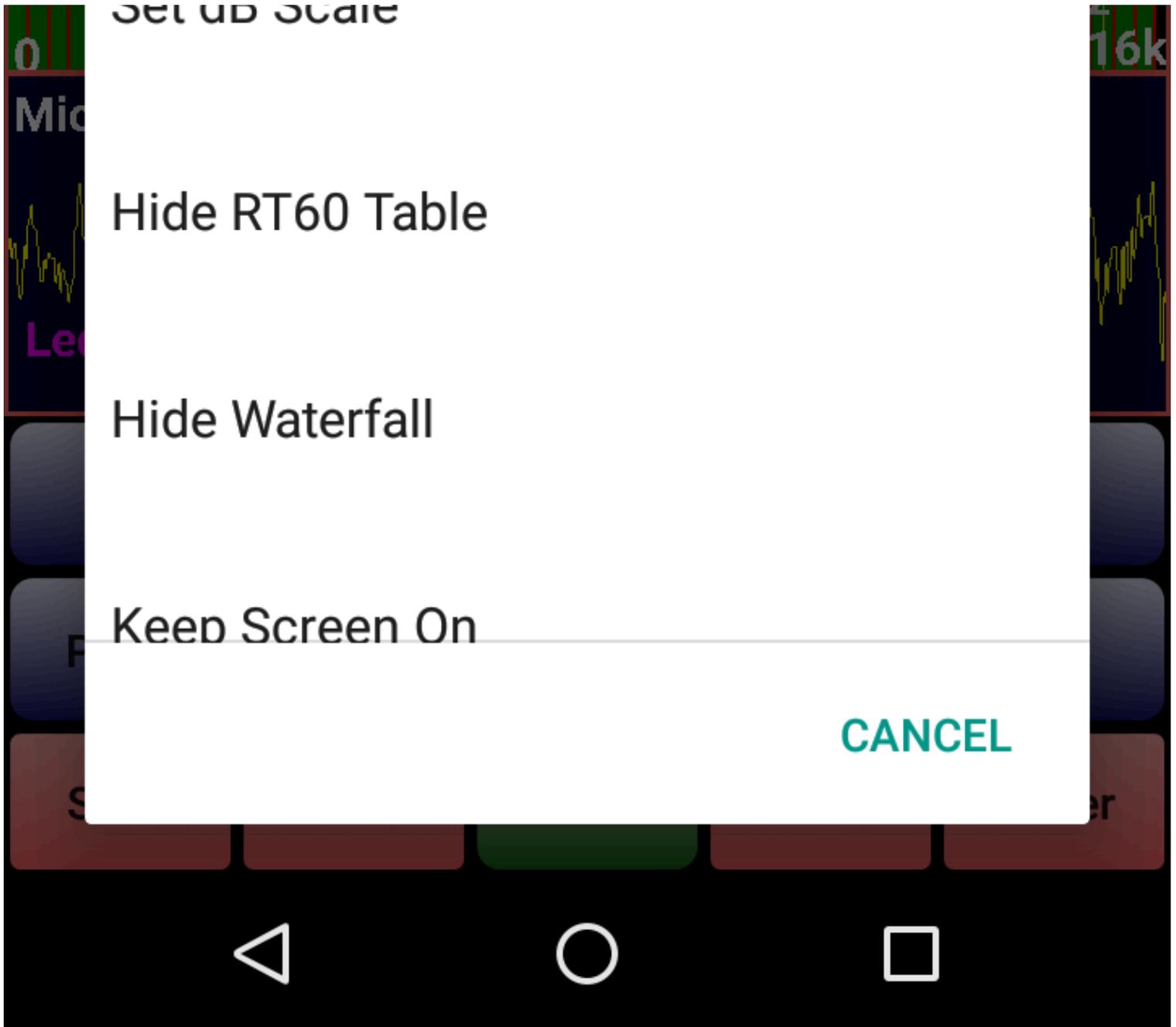


You can store a Peak Store measurement into a file, by using the "Store" button. To compare a previous measurement with the current, use the Load button to select the previously stored measurement file. In the screenshot above, a previously made measurement called "ambient" is being compared.

## Scales

In the RTA and Spectrogram modes, the user may choose to have AudioTool use a logarithmic or a linear scale for the frequency (Hz) axis. The choice is made via the Menu:





Choosing "Set Lin Hz" menu option will change to the linear scale.

The frequency limits for the Hz scale can be adjusted, using the menu option "Set Freq Scale". The following dialog will appear:



**Set Freq Scale Limits (Hz)**

Minimum freq(>20.0), Maximum freq(<23000.0)

20.0    10000.0

OK    Use Default    CANCEL

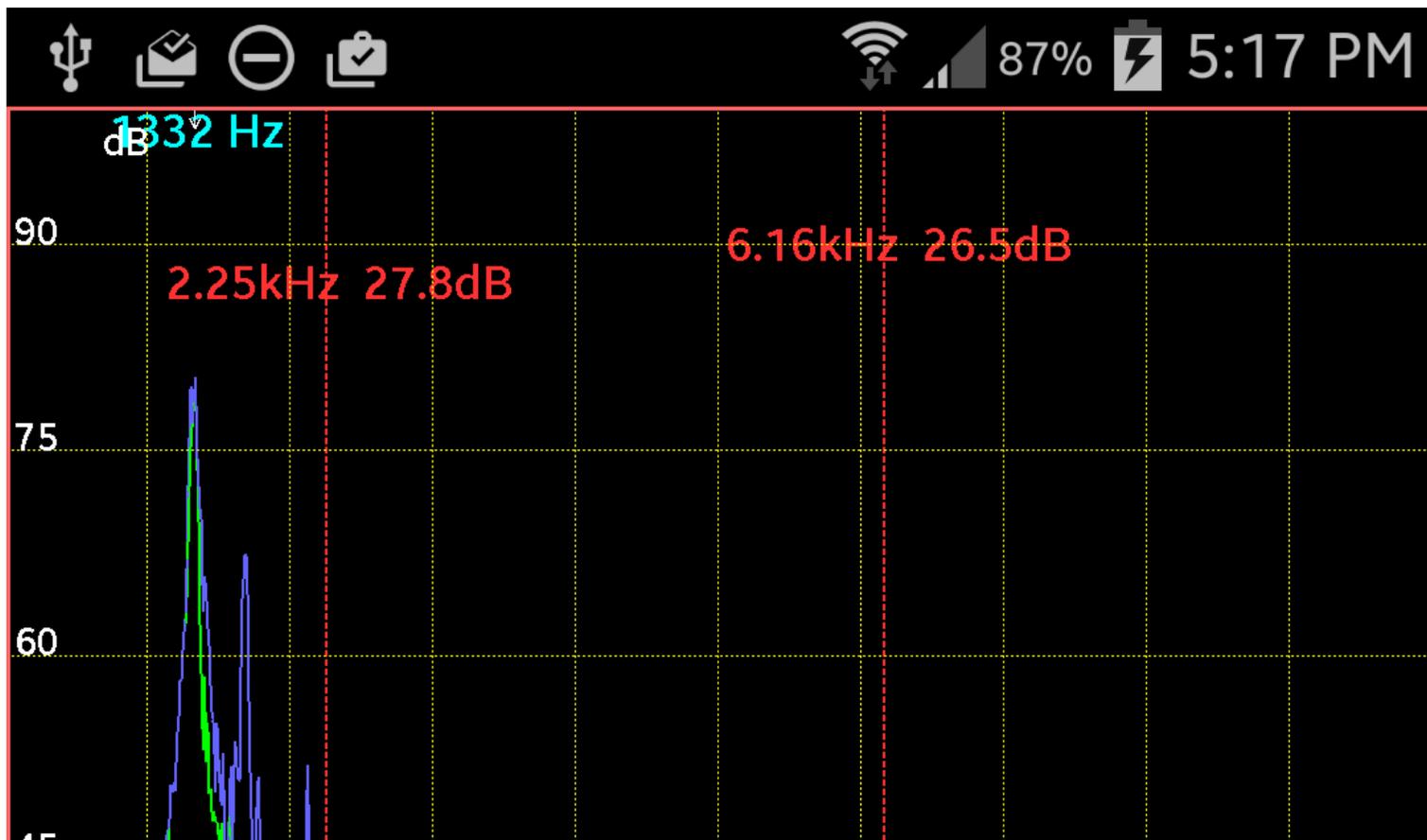
100 00    10 000490621

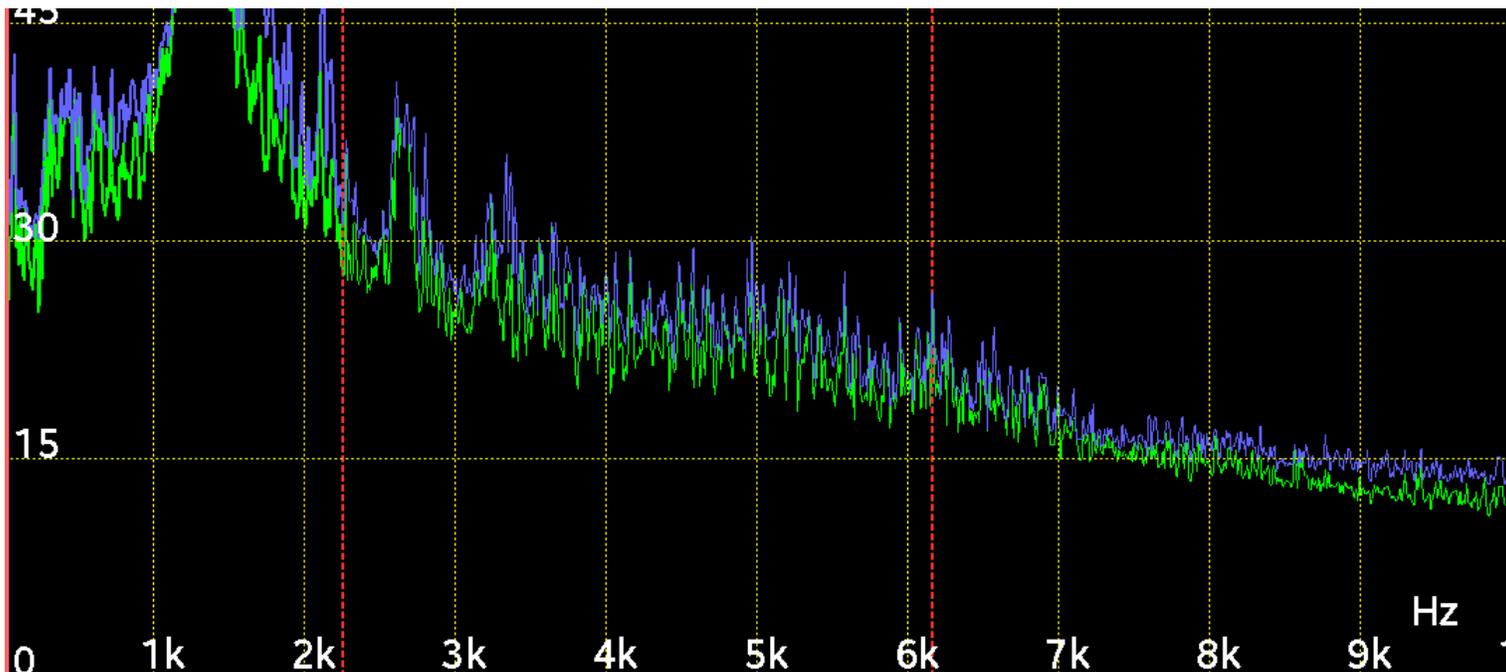
1 2 3 4 5 6 7 8 9 0

q w e r t y u i o p



In the example screenshot above, the user has set limits of 20Hz to 10kHz. Note that you can set the scale limits back to their default values by tapping the "Use Default" button. Here is what the linear scale looks like:





Mic 44100 16 8192 dF=5Hz C=0.0

90 dB SPL

Running

Average Off

Full Res

Peak On

Vall Off

Fast

Flat

Store

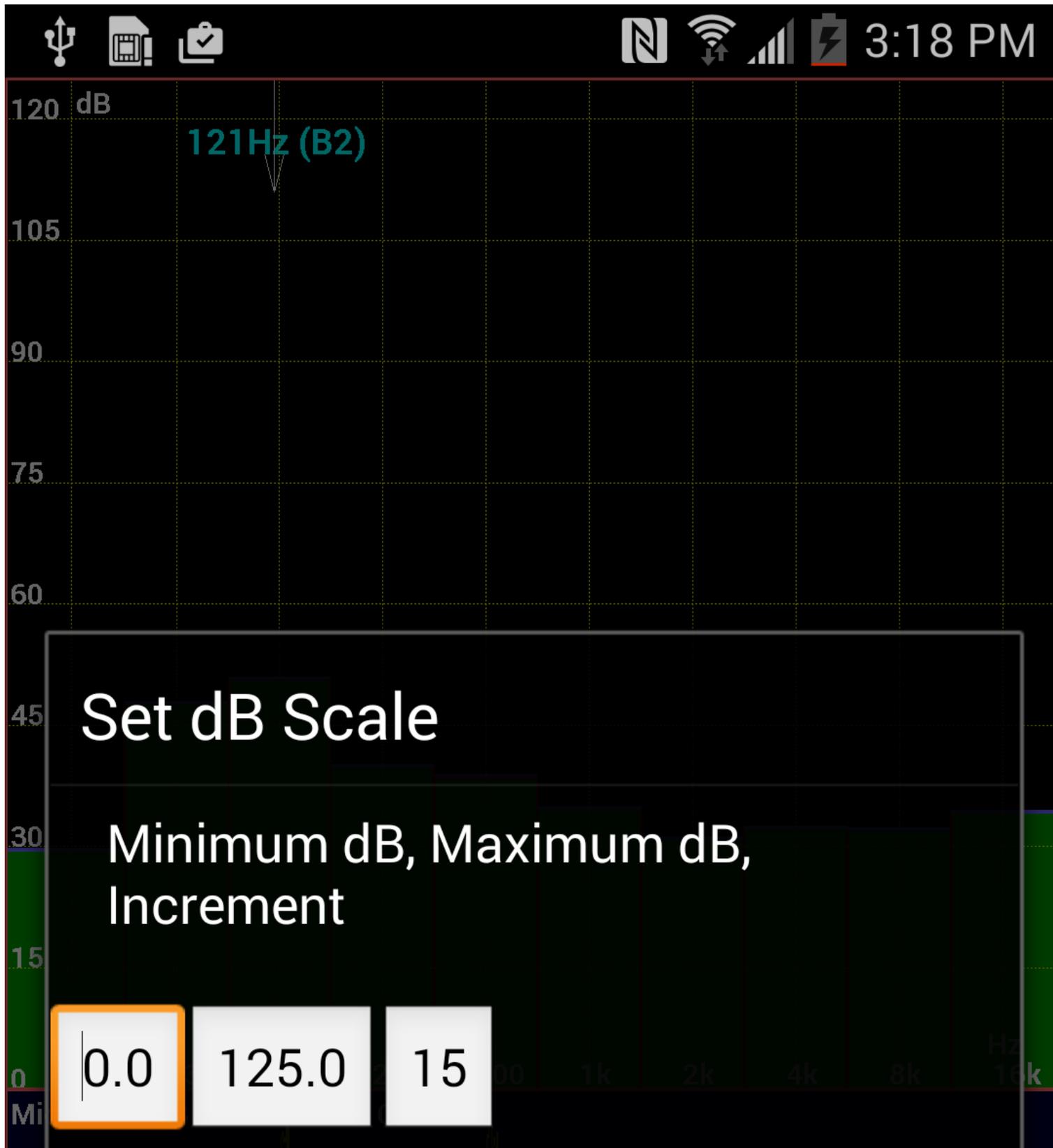
Load

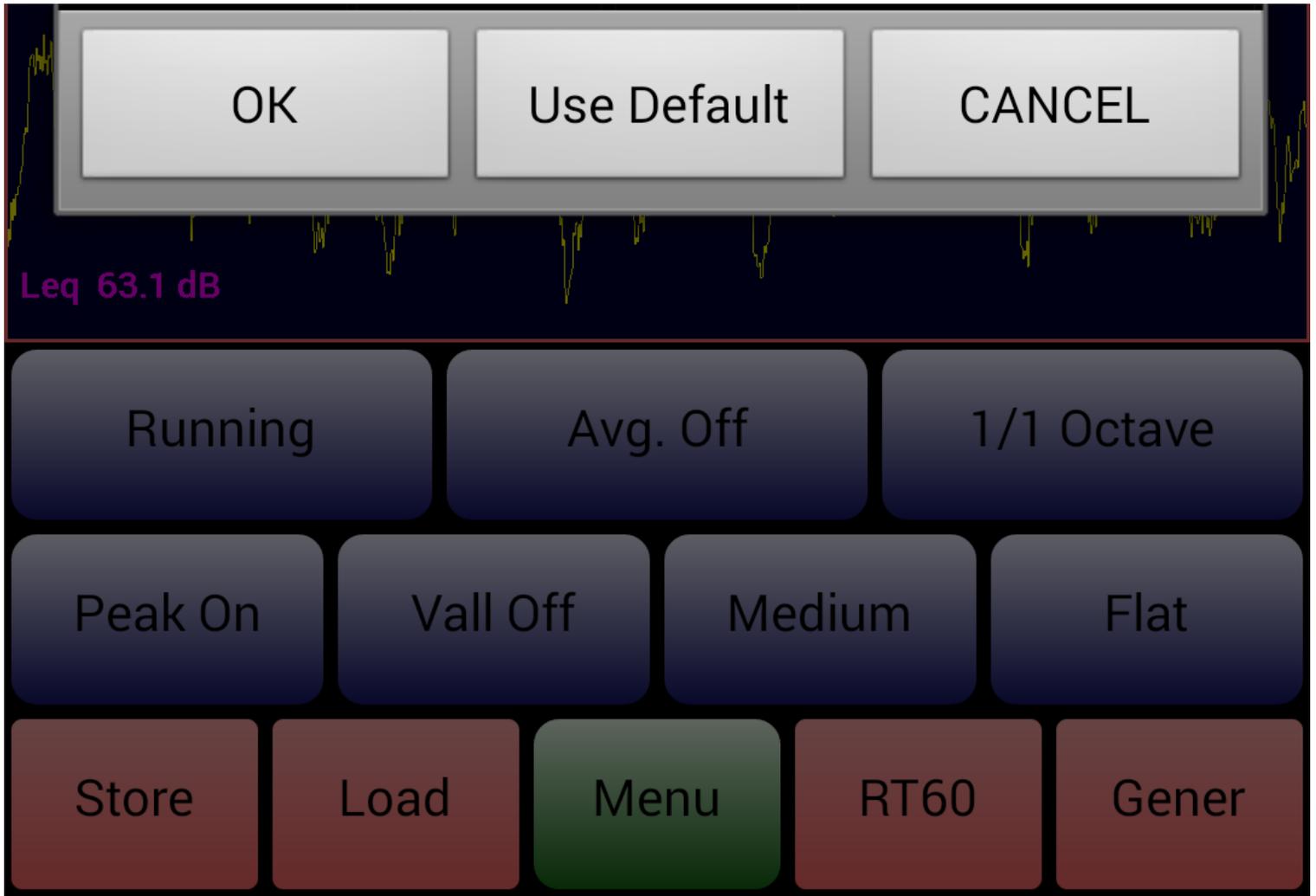
Menu

RT60

Gener

You can similarly adjust the dB Scale limits, via the Menu. The "Set dB Scale" Menu option also allows you to specify the increment between scale lines (default is 15dB):





Note that the Hz scale limits, and the dB scale limits and increment, are **saved** by AudioTool when the user terminates the app: they are **restored** when the user restarts AudioTool.

### Scroll and Pinch to Adjust Scales

You can adjust the dB and frequency markings in the RTA and the dB marks in the Chart Recorder in the following way:

- By dragging your finger up and down the screen, the dB centre line is adjusted
- By dragging your finger left and right, the frequency centre line is adjusted (not in Chart Recorder)
- By pinching (with two fingers) inwards or outwards upwards, the dB scale is adjusted
- By pinching inwards or outwards left and right, the frequency scale is adjusted

(Scroll and Pinch of the scales in this way is disabled when the Cursors are visible.)

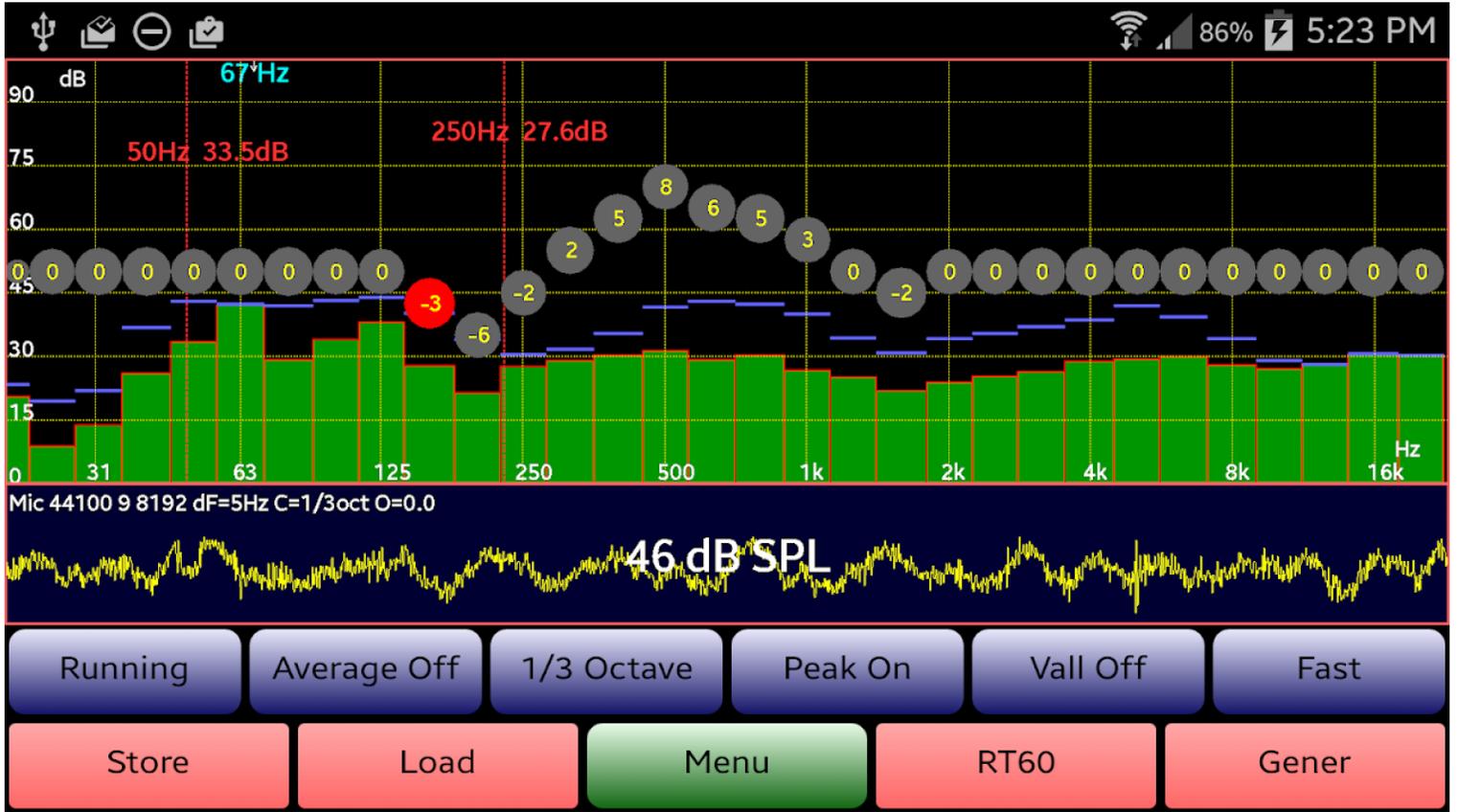
Again, note that you can set the scale limits back to their default values by tapping the "Use Default" button.

## Calibration

There are two independent methods of Calibration provided with AudioTool. You can select which method to use in the Menu. The first is a "single value" method where you can enter a single number to offset the broadband SPL measurement so that it matches an external SPL meter. This is useful for simplicity. The second is an ISO 1/3 Octave Calibration where you can correct for the response of the phone's microphone in each of 31 1/3 octave bands. In either case, to calibrate you will select "Calibrate" from the Menu.

AudioTool will store and remember the calibration number(s) you enter, and will also remember whether you are using the single value or octave based method.

For the 1/3 octave based method you will see a screen similar to that shown below:



Each of the 1/3 octave band's current calibration values (decibels correction, positive or negative) is shown in a grey disc - when you first calibrate, all values will be zero. The red disc shows the band currently being adjusted. If you double tap on the left hand side of the screen, the selected disc moves to the left. If you double tap on the right side of the screen, it moves to the right.

To adjust the selected band, place your finger anywhere on the screen and drag the up or down - the selected disc will move in unison, until you see the correction value you want. Take your finger off the screen when you have adjusted the disc, and then double tap to select the next band, and so on.

Your aim is probably to move the discs so as to obtain a flat response, and match the overall SPL to that of an external meter, or to correct for any odd peaks or troughs your Android phone may exhibit in its frequency response.

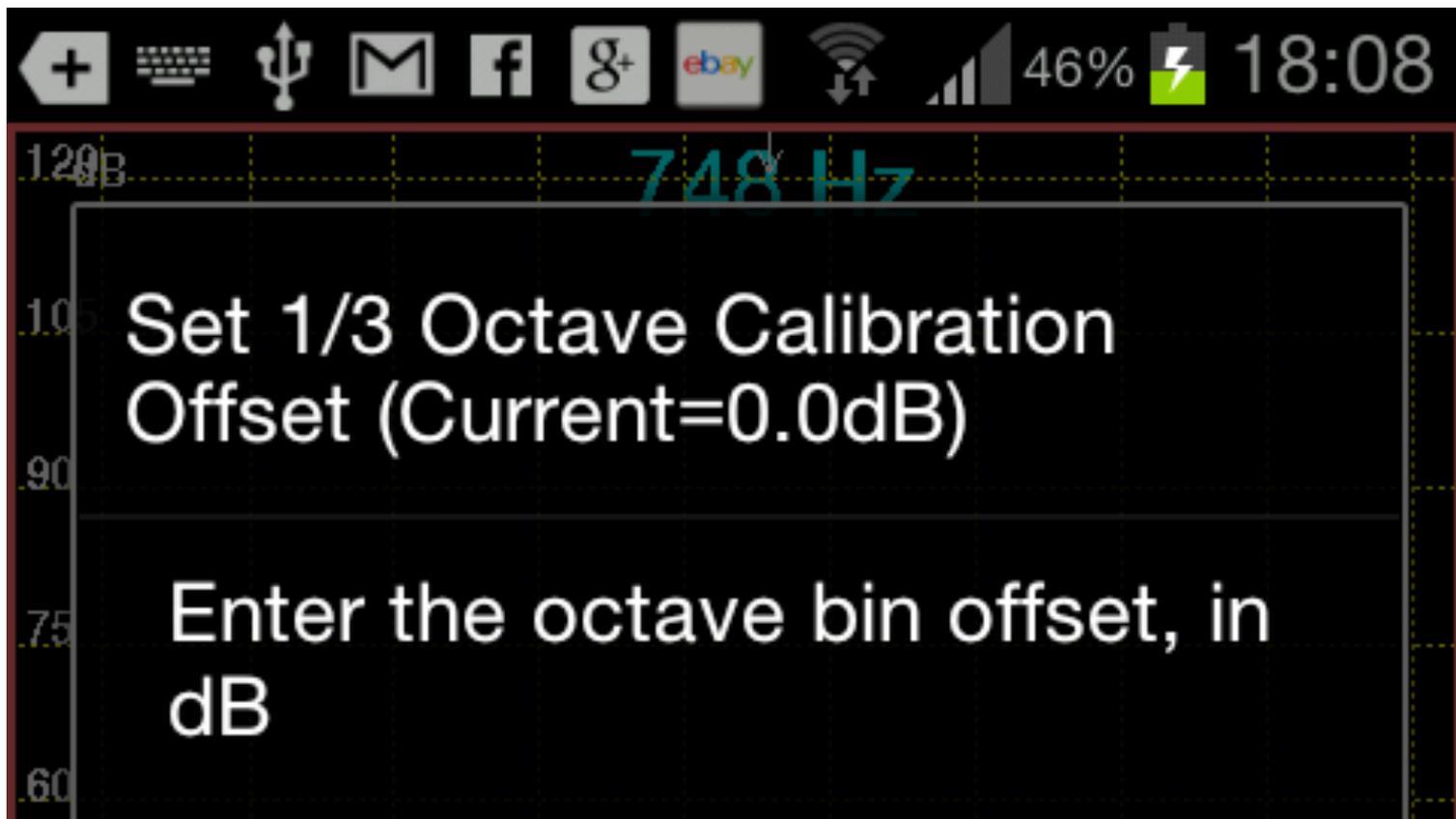
When satisfied with the results, choose "Finish Calibration" from the Menu, and confirm whether you want to save the values. These values will be retained on your phone and will be loaded every time AudioTool starts.

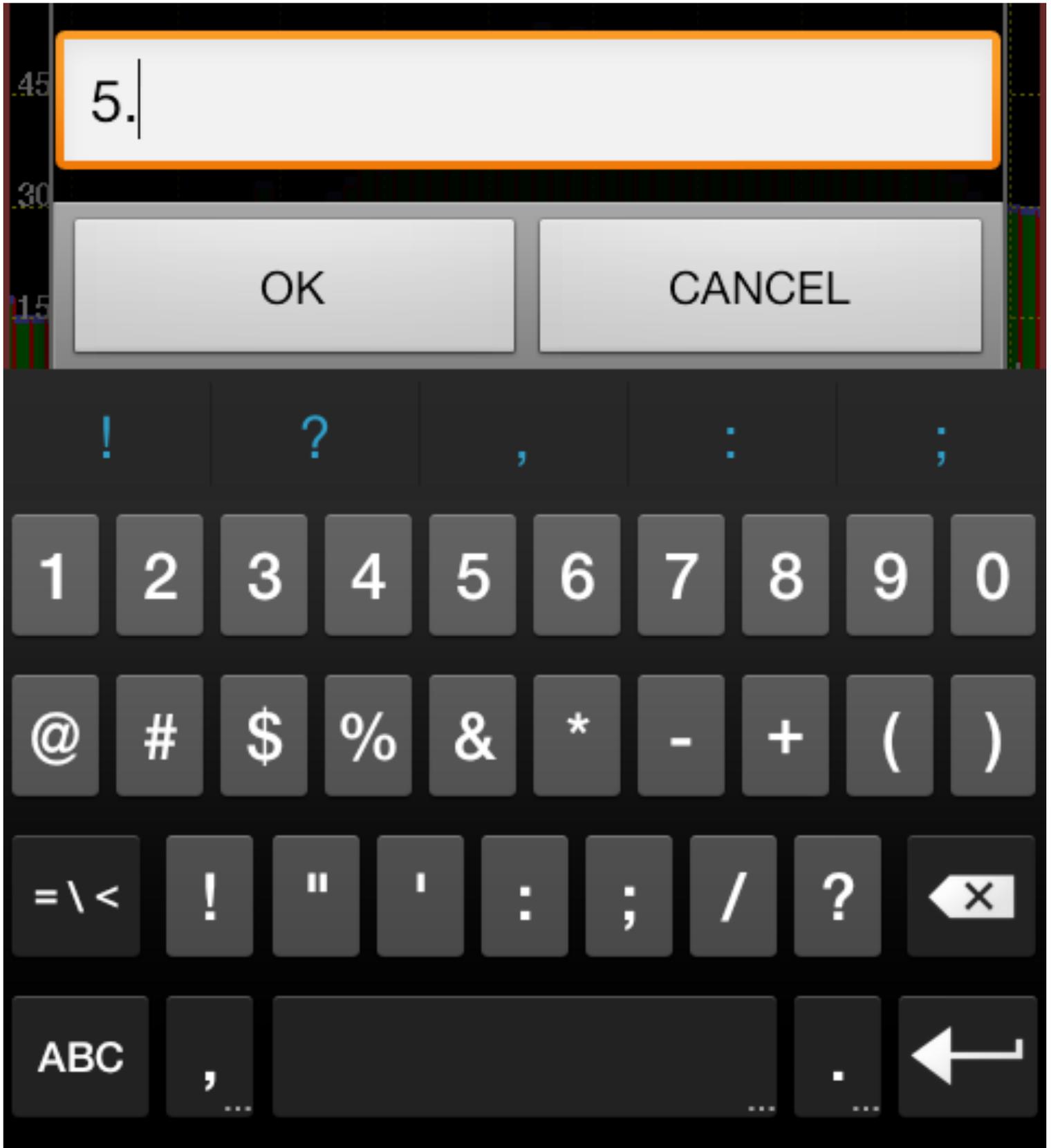
### Global Offset

By default, the dB calibration values in each band are all 0dB. You can apply a global offset to these values using the "Set Cal. Offset" Menu option (which appears when you are using the 1/3 Octave calibration method.)

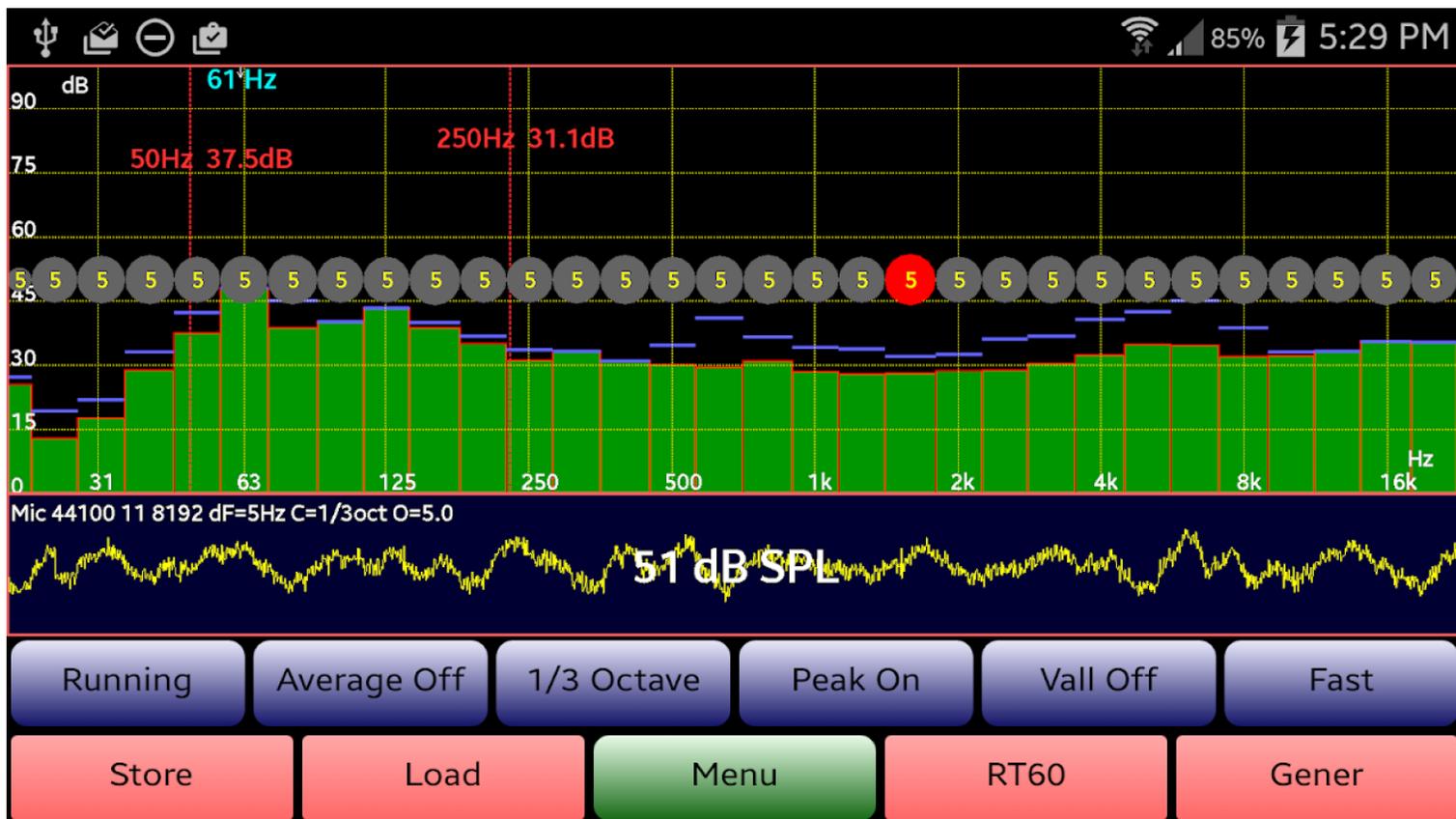


The "Set Cal. Offset" option will bring up a dialog where you can enter the desired value:





Now, when calibrating, the dB levels in all bands will be offset by the amount you entered:



You can adjust the levels in each band in the way described above. Remember to select "Finish Calibration" from the menu when you are done!

(The observant reader will notice that the current global offset value is handily shown in the Mic display, as O=5.0)

### Dayton Audio Calibration Files

AudioTool supports loading a calibration file for the [Dayton Audio iMM-6 microphone](#). These files have names like "99-0101.txt" when downloaded from Dayton's website. For use in AudioTool, you will simply need to rename the file so that it has the .cal filetype (e.g. "99-0101.cal") and save it in the AudioTool directory on your Android phone. Then, in AudioTool Menu, click on "Use 1/3 Octave Calibration" to ensure you have the 1/3 octave calibration method selected. You can then select the "Load Cal" option from the Menu, and choose the Dayton file from the list. The calibration data will be loaded, power summed to the 1/3 octave bins used by AudioTool, and saved in your Preferences - there is no further need to load the file whenever you start AudioTool, unless you change the calibration method or alter the calibration values manually.

Please see the section above on "Global Offset". Just because you have an iMM-6 and have loaded the Dayton cal file, does *not* mean that AudioTool will necessarily agree with an independent pro

Dayton cal file, does not mean that AudioTool will necessarily agree with an independent pro-grade meter. To get agreement, you should adjust the Global Offset. Even then, if the internal audio circuitry on your device is not flat, your results will vary.

### MiniDSP PMIK microphone Calibration Files

Download the calibration file for your PMIK mic from MiniDSP, and rename it so that it has the .cal filetype, similarly to the procedure described in the above section for the Dayton mic.

If you are running AudioTool version 7.4 or earlier, you will need to edit the .cal file, using a text editor. The trick is to add a "\*" at the start of the first line, which tricks AudioTool into thinking it is a Dayton cal file. E.g. if the first line in the PMIK file looks like this:

```
"Sens Factor =-18.1125dB, SERNO: 7123456"
```

change it to this:

```
*"Sens Factor =-18.1125dB, SERNO: 7123456"
```

Save the modified version, and Load Cal it into AudioTool. Refer to the instructions above for Dayton files for further information.

### Rex Beckett's Calibration Method

Rex has provided a handy Excel tool to assist in calibrating your Android device. This tool is now at version 2.0. Rex says: *"I have updated the Excel tool for preparing AudioTool calibration files. The new version uses the 1/3rd Octave data that Julian has helpfully appended to the captured spectrum (at) files. This allows it to work with files captured with FFT resolutions of 4096, 8192 or even 16384 (fine). It is also much faster! As well as processing the captured pink response, the tool can load calibration files for Dayton iMM6 and miniDSP UMIK-1 measurement microphones. It can also be used for manual entry of microphone response factors. The process is essentially the same as that described in the first post except that microphone responses are managed on dedicated worksheets. Once the microphone response has been loaded or entered, it can be sent to the Main sheet by clicking the Send to Main button. In this way, cal files for several different mics can be prepared for the same pink response.*

*I think this tool should be able to cope with at files produced by recent versions of AudioTool."*

### Rex Beckett's AudioTool to REW Converter

Rex also provides an Excel tool for converting AudioTool .at files into REW format. He says *"The tool*

-----  
 can load AudioTool at files, extract the 1/3rd octave data and show it as both a raw plot and as a response relative to 1kHz. It can then make a tab-delimited txt file that can be imported as a measurement by REW. The REW file may be made from either the Raw or Relative response as selected by the drop-down control."

## Target Curves

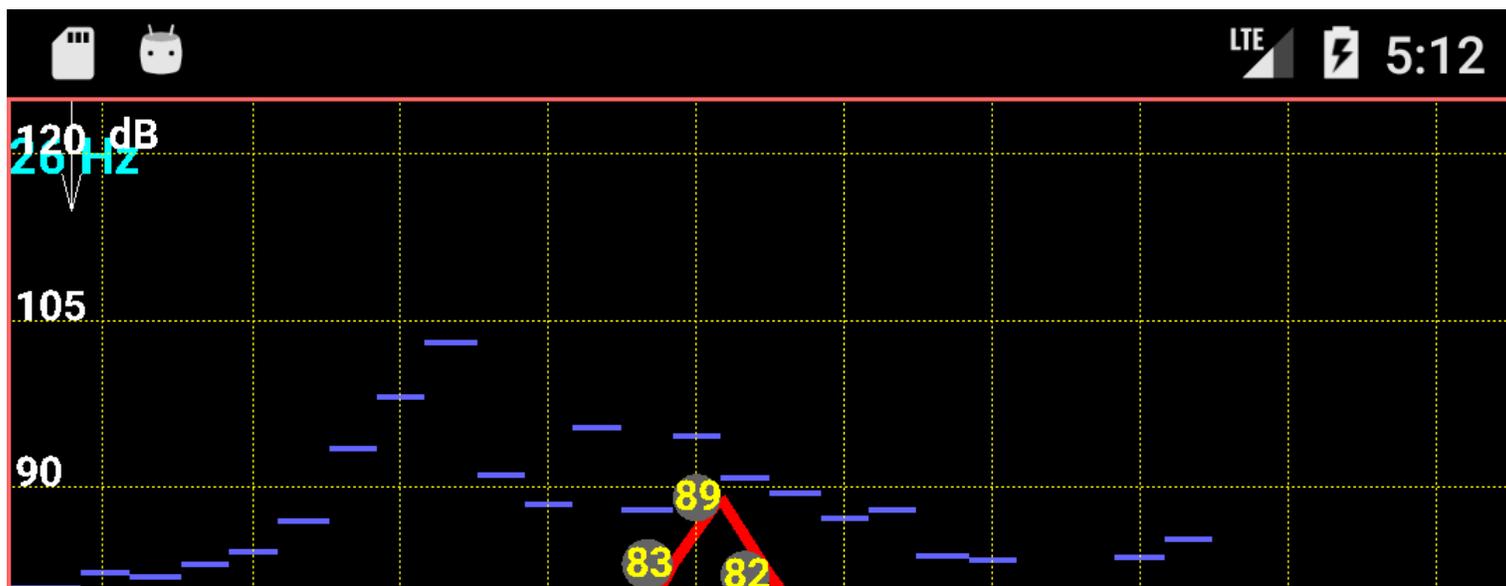
A target curve can be used as a visual guide when adjusting levels of an audio system to match a required shape, or for comparing a frequency response to a template, for examples. The target curve is shown as a red line across the screen in the RTA Octave modes. It can be any shape you like: the default target curve is a straight line at 60dB SPL.

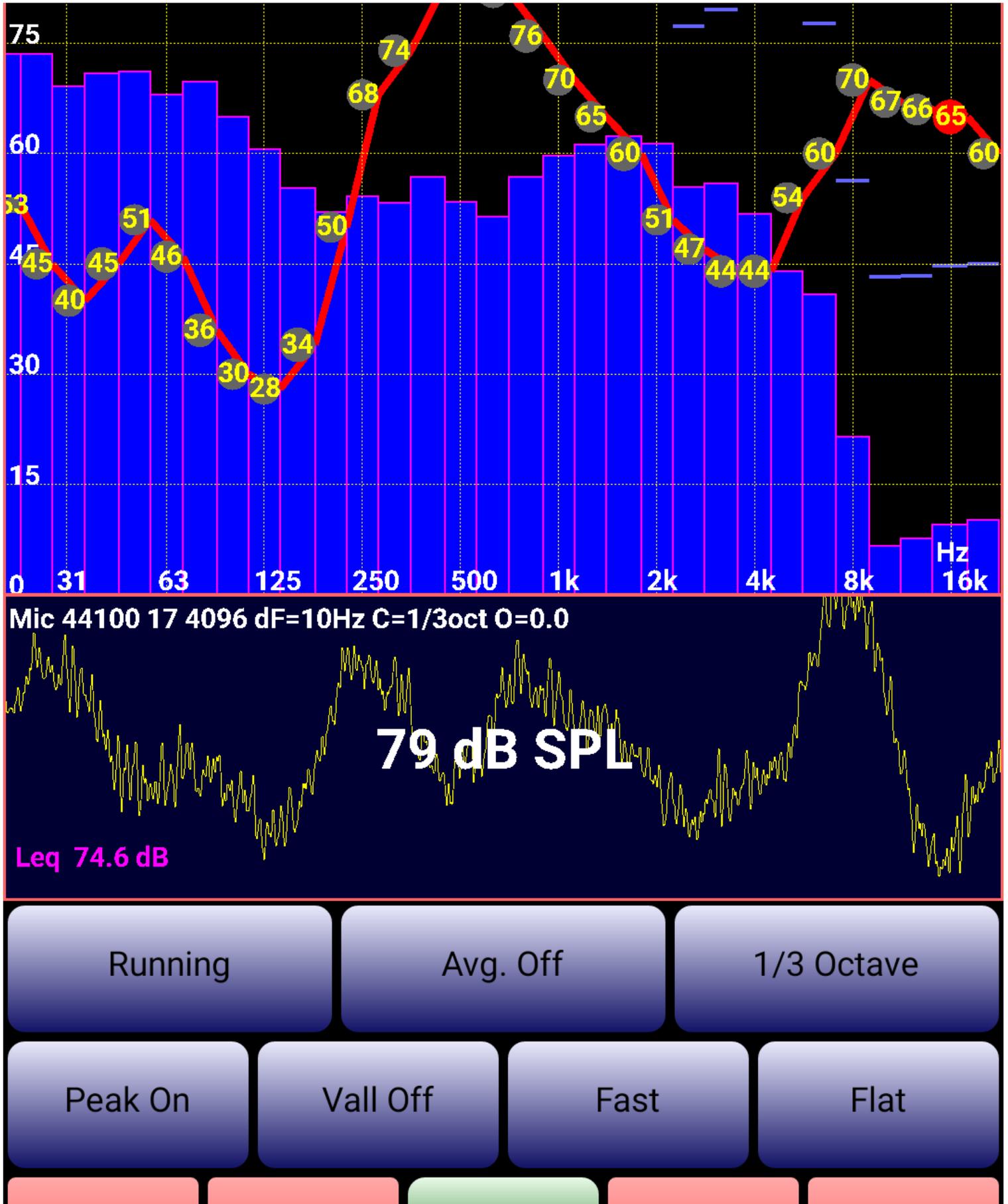
To set a Target Curve, enter 1/3 Octave RTA mode, and use the Menu option "Set Target Curve". (You can be in other RTA modes to do this, but the Target Curve is only defined at 1/3 Octave resolution).

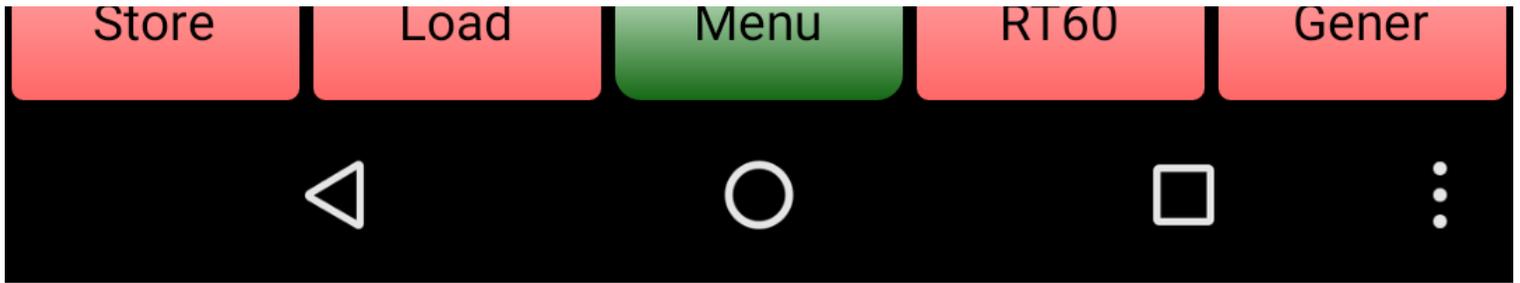
You will be shown a set of discs with numbers in them (this is very like AudioTool's 1/3 Octave Calibration scheme for setting calibration levels).

Initially, the active disk is the middle of the frequency spectrum: the central 1/3 octave band, and is shown in red. Slide your finger up the screen to increase the dB level of the disc, and down the screen to decrease it. Once you have set the level for that band/disc, double tap on the right hand side of the screen to move to the next band/disc (it will become red), and proceed like that until all levels are set. Double tap on the left hand side of the screen to move to lower bands/discs.

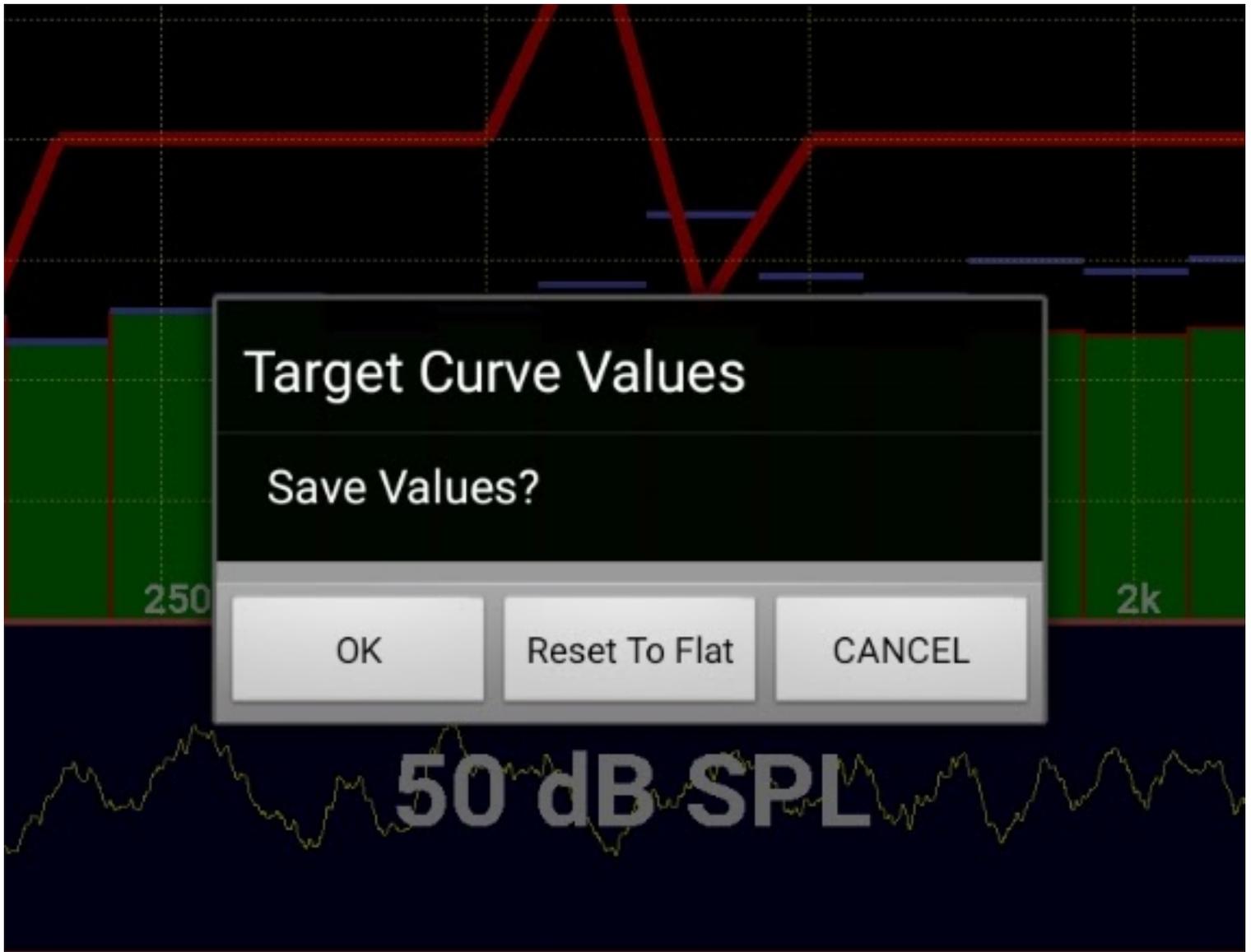
Here is an exaggerated example of a completed target curve:







Once you are happy with the Target Curve, enter the Menu, and select "Finish Target Curve". You will be prompted to either save the values, reset the values to flat, or cancel.



Make sure that, in the Menu, you have "Show Target Curve" selected, if you want to see your Target Curve. The Target Curve values you save will be reloaded every time you restart AudioTool.

If you would like to save different types of Target Curve, you can do so via the menu's "Save Target Curve" and "Load Target Curve" options. This allows you to save and load Target Curves that are used for different purposes.

If you would like to create your own Target Curve files, this can be done using a text editor: create a file that has the following format:

```
AudioTool Target Curve Comment
```

```
20 60
```

```
25 60
```

```
31 60
```

```
40 60
```

```
50 60
```

```
63 60
```

```
80 60
```

```
100 60
```

```
125 60
```

```
160 60
```

```
200 60
```

```
250 60
```

```
315 60
```

```
400 60
```

```
500 60
```

```
630 60
```

```
800 60
```

```
1000 60
```

```
1250 60
```

```
1600 60
```

```
2000 60
```

```
2500 60
```

```
3150 60
```

```
4000 60
```

```
5000 60
```

```
6300 60
```

```
8000 60
```

```
10000 60
```

```
12500 60
```

```
16000 60
```

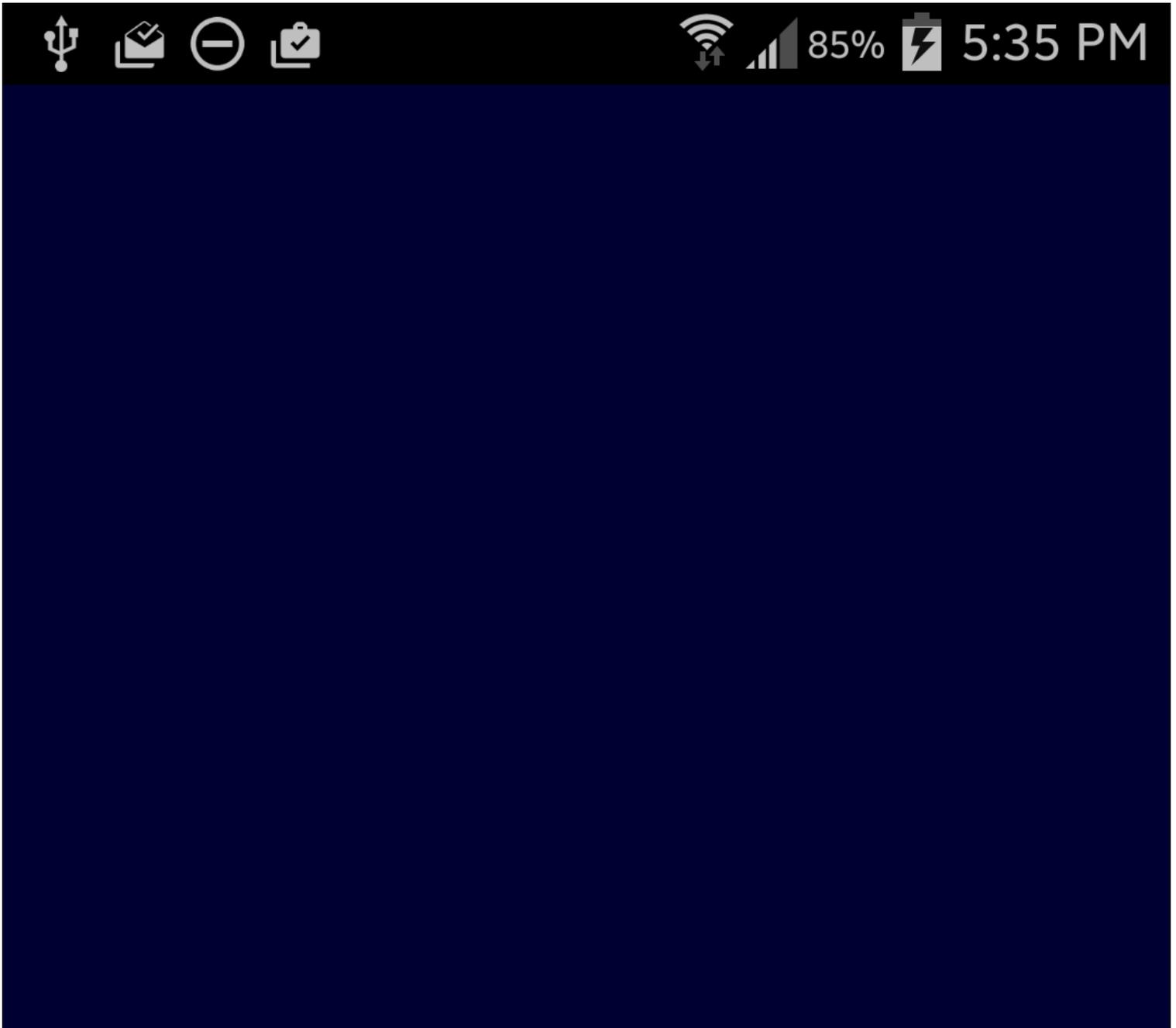
```
20000 60
```

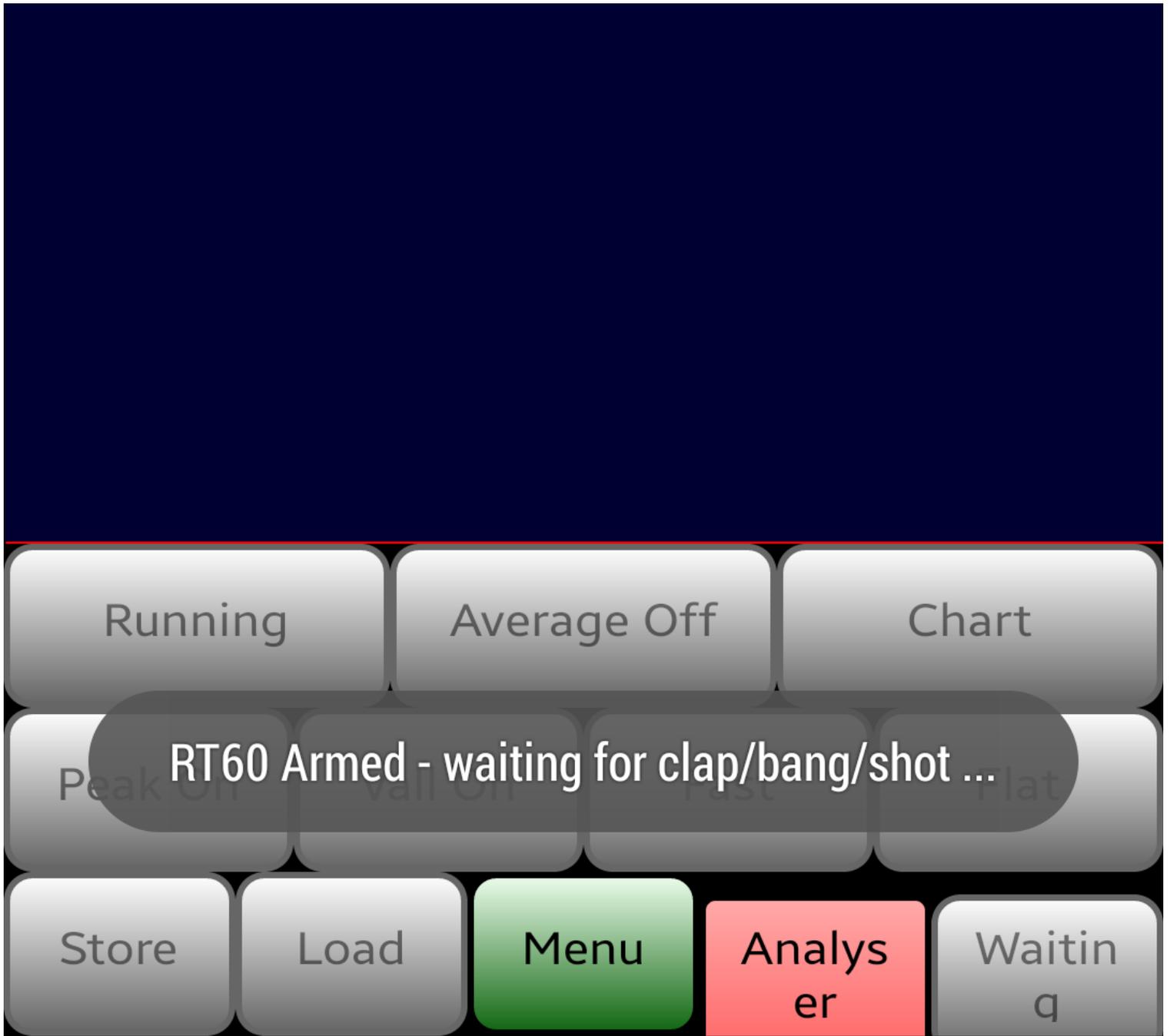
(Each row after the first has the frequency, followed by the Target Curve level in dB ... in the example above, all 1/3 Octave bins have a level of 60dB.)

Or, you could save a Target Curve using the "Store Target Curve" Menu option, and edit the file so created. Target Curve files have a filetype of .targ.

## RT60

You can measure room reverberation times in RT60 mode, selected using the RT60 button.

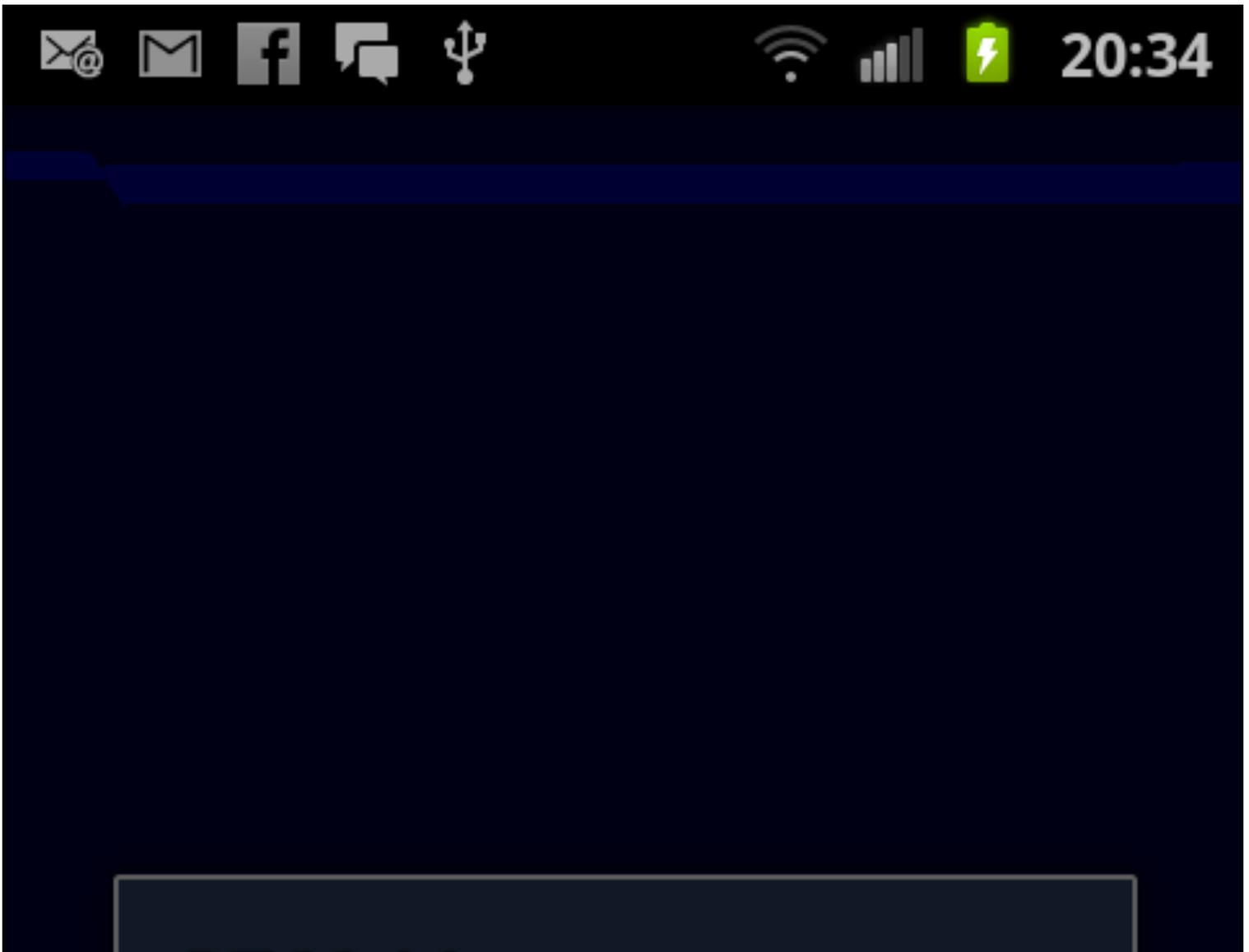




If the Signal Generator is Off when RT60 mode is started, the system waits for a loud impulse signal, such as a clap to trigger the measurement. This sound will energise the room, and how quickly the sound pressure falls off with time will be measured. The longer the sound takes to die away, the more reverberant is the room.

After the clap is detected, the measurement starts. Since the RT60/reverberation time for a room will be different depending on frequency, AudioTool makes a wideband (all frequencies) measurement, as well as measurements in each of the ISO octave bins. To do this, the data are passed through a bank of IIR octave filters centred on the desired frequencies. The results are displayed in tabular form, and may be exported to a file using the "Store" button.

RT60 measurement can also be made using the signals (e.g. White Noise) from the generator to energise the room. If the Signal Generator is On when RT60 mode is started, then the Trigger button:



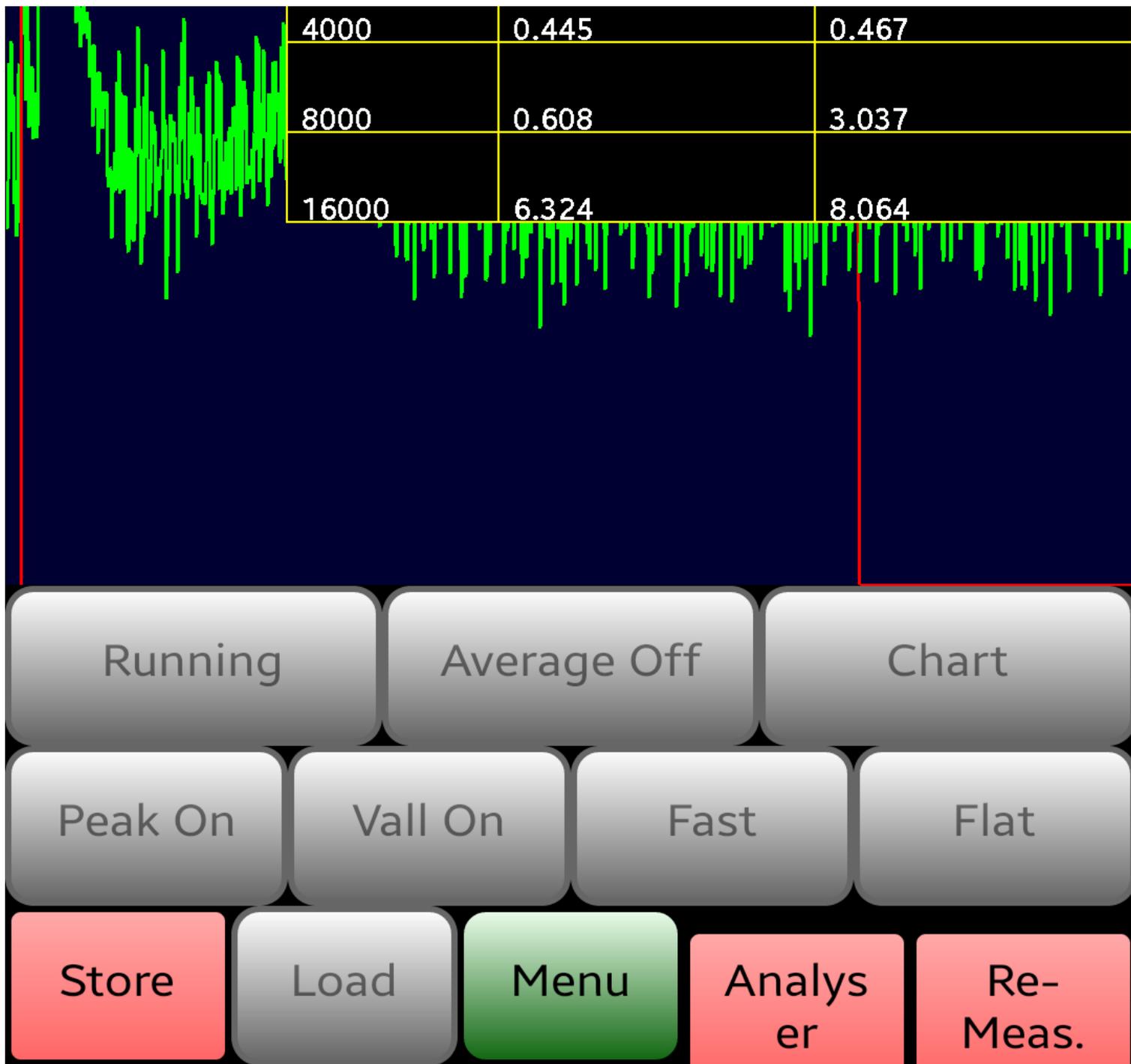


can be pressed when the RT60 measurement should be made. When the Trigger button is pressed, the output from the Signal Generator is stopped, and the RT60 measurement of the decaying sound intensity is made.

Technical notes: the RT60 uses a Schroeder integral to work backwards from the noise floor to the detected peak intensity. The Schroeder integral is shown as a red trace on the RT60 display. The RT Early number is the time for the intensity to drop from the peak to 10dB below the peak, multiplied by 6. The 15-20 number is the time for the intensity to drop from 5dB below the peak to 20dB below the peak, multiplied by 4.

The screen shot below shows an example of an RT60 measurement.

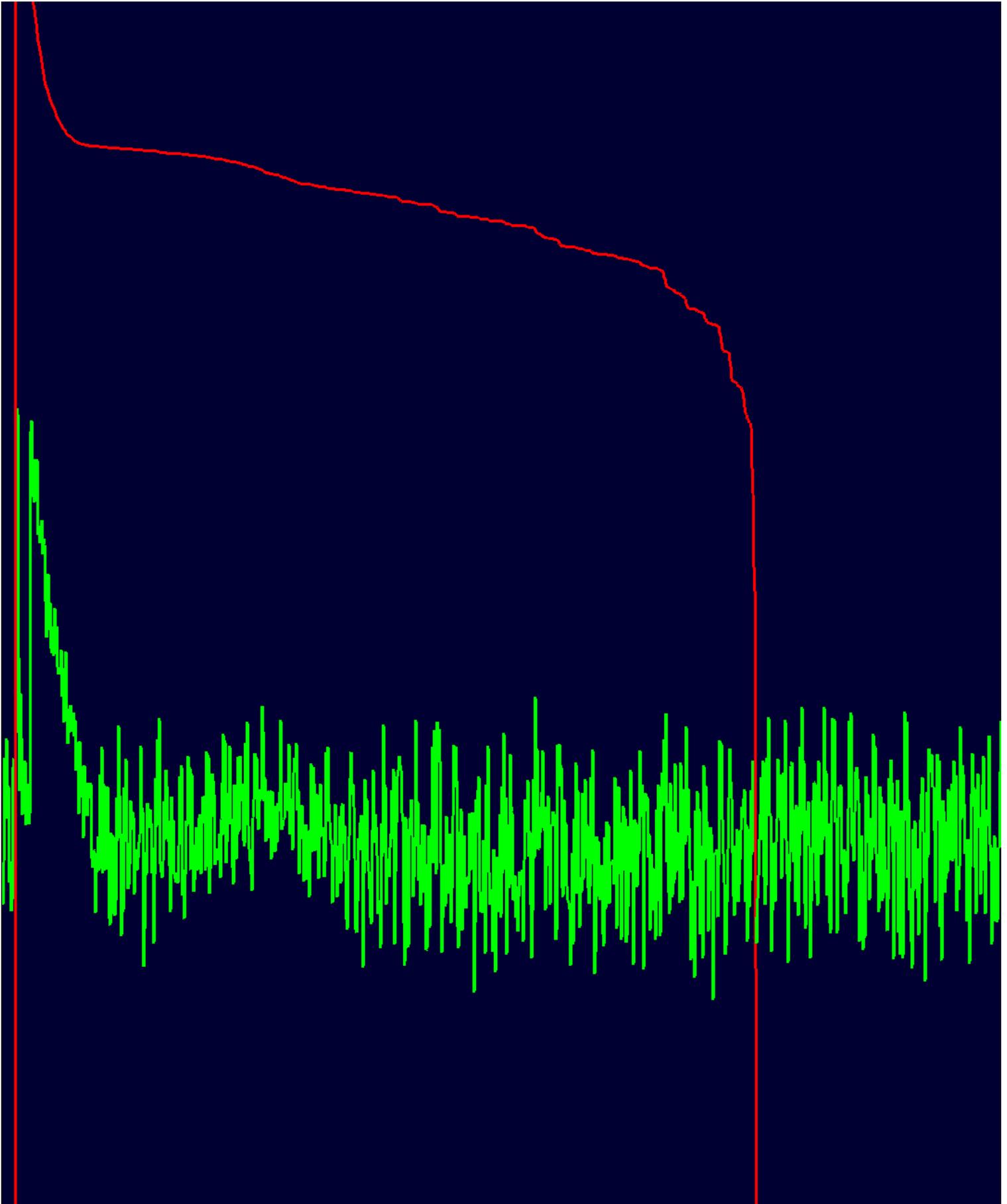


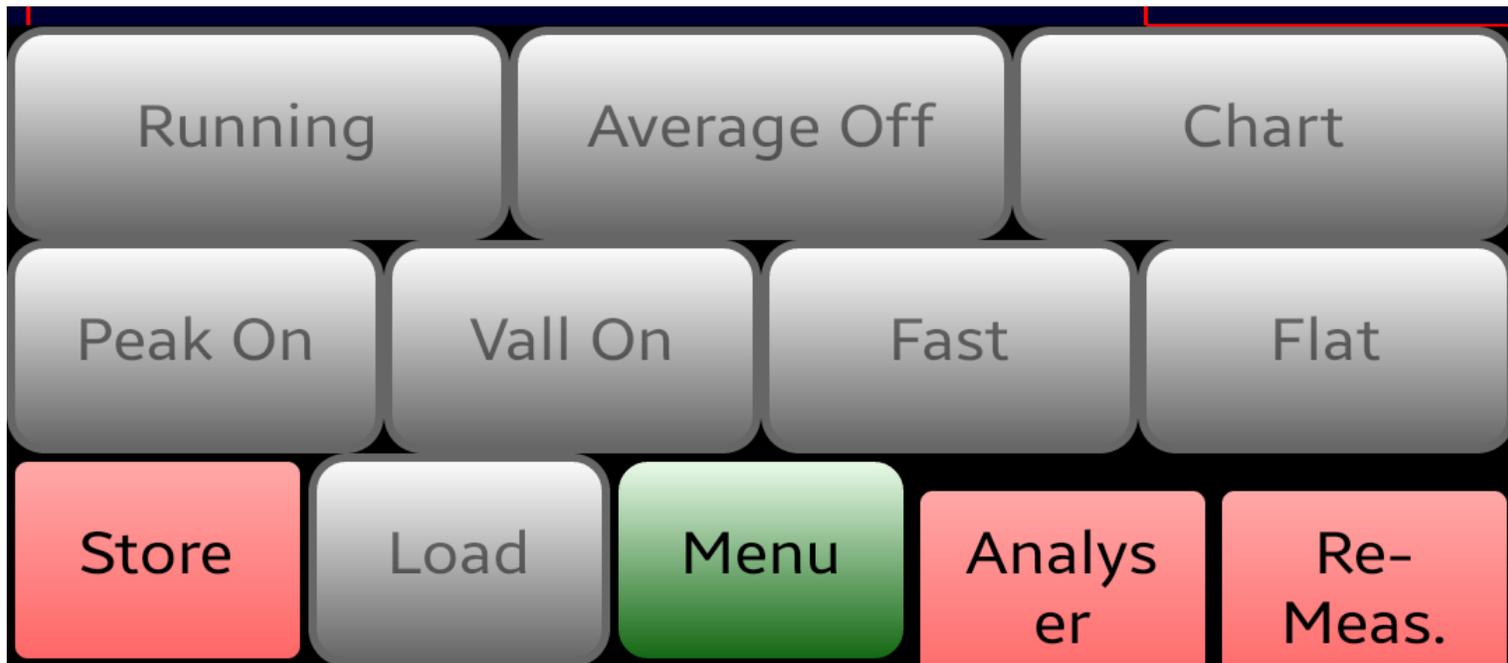


The RT60 results table can be hidden or shown by using the Menu option "Show(Hide) RT60 Table".

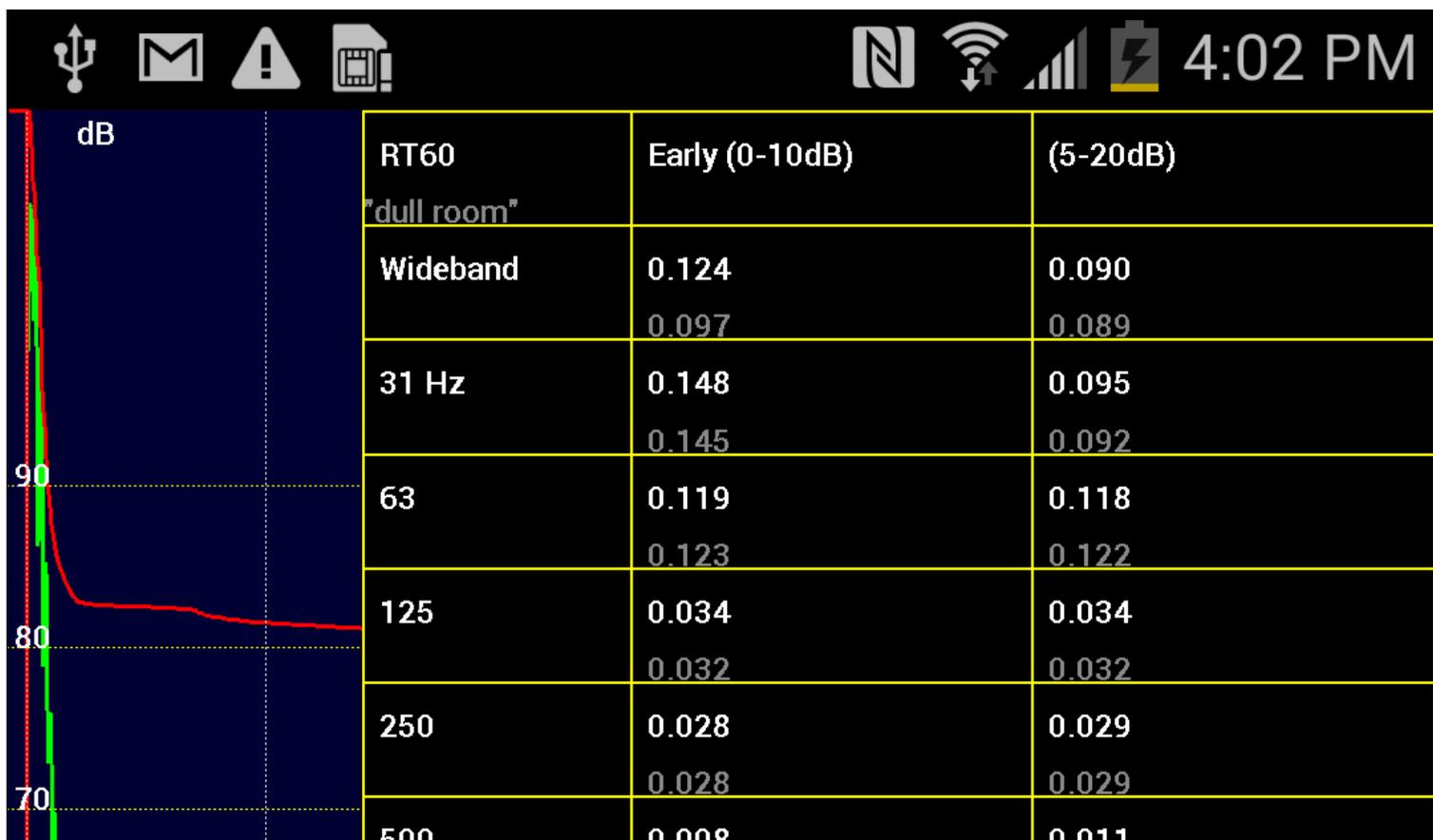
Hiding the RT60 table allows closer inspection of the RT60 curve data:







You can also Load a previously Stored RT60 file, to compare measurements, using the Load button. The measurements from the file are then shown alongside your latest measurement in the RT60 table. The example below shows a table with a new measurement compared with that loaded from a previously stored "dull-room" RT60 file.

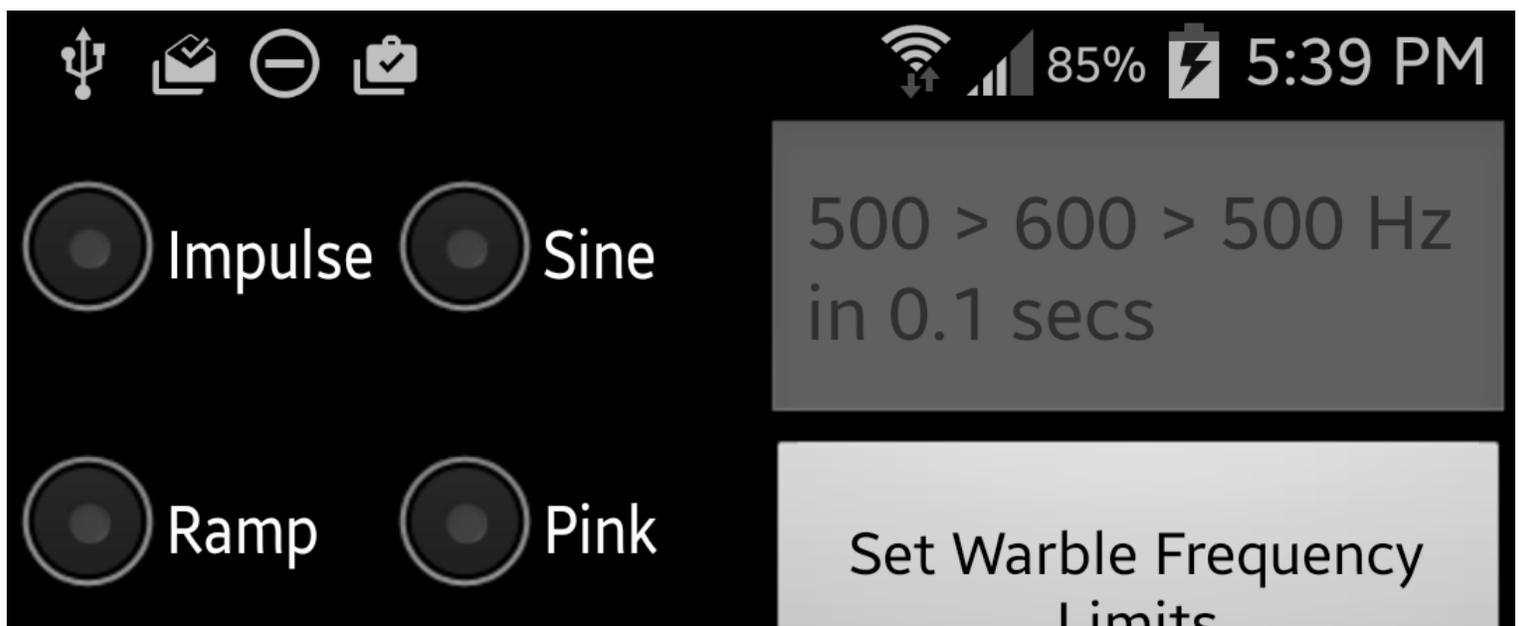


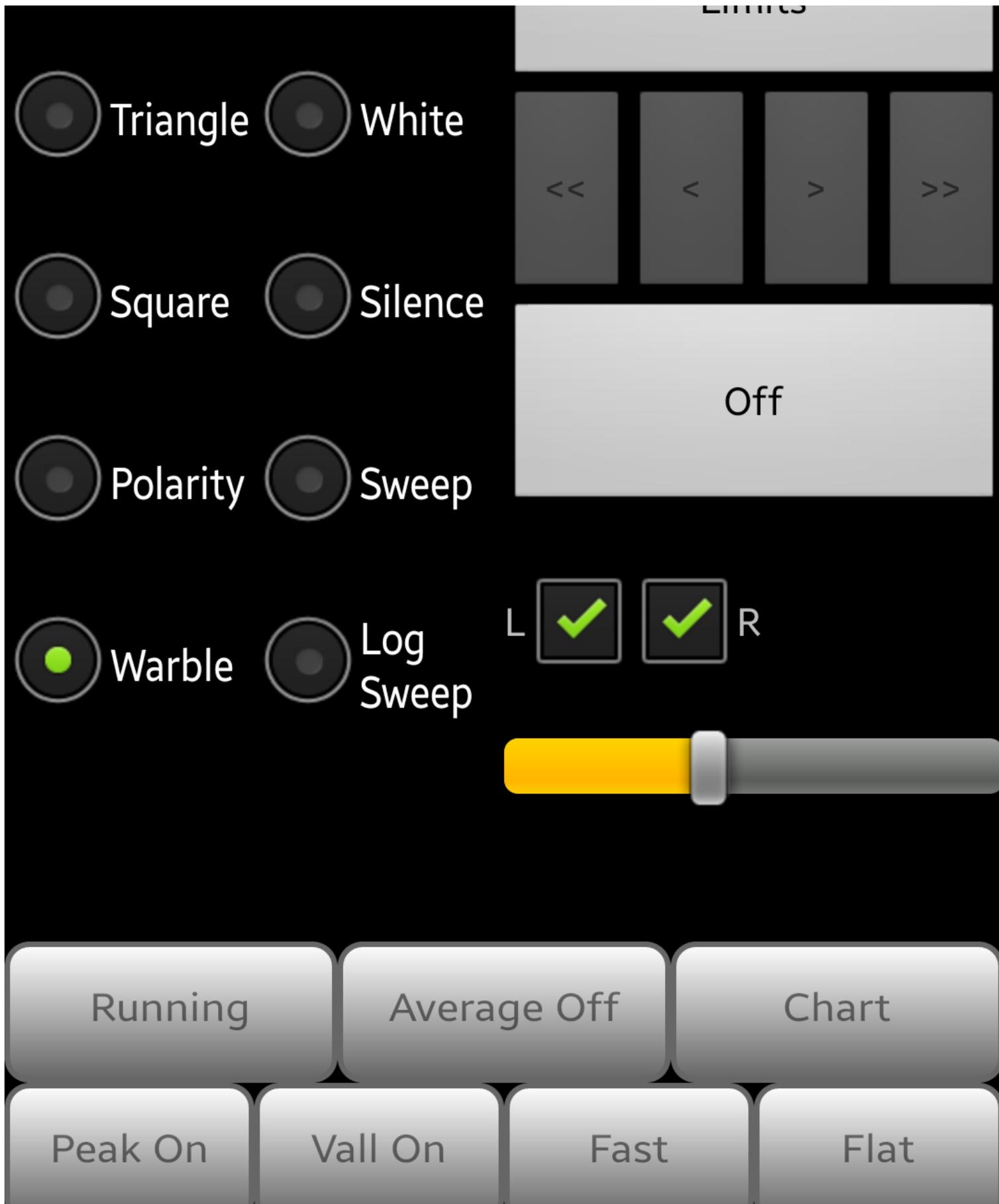


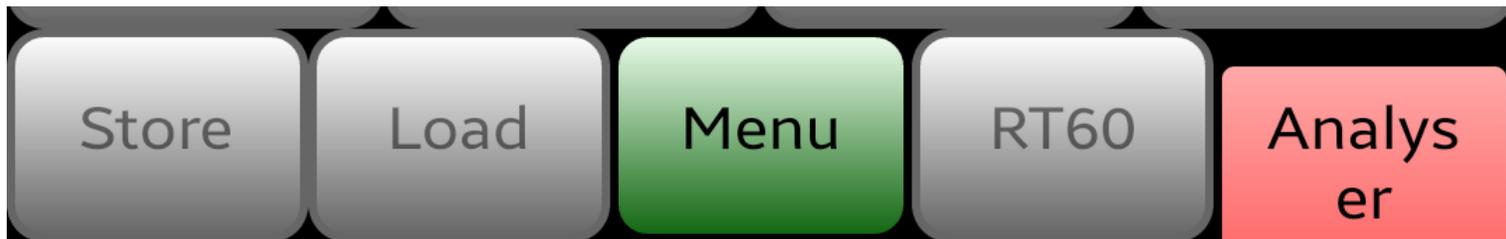
# Signal Generator

The Signal Generator allows you to produce several different types of waveforms from the phone/tablet's audio subsystem.

- Impulse: this is a periodic pulse that is produced once every second or so
- Ramp: this is a waveform which ramps from negative to positive, then drops to negative - also called a sawtooth
- Triangle: this waveform is triangular in shape
- Square: a rectangular waveform with 1:1 duty cycle
- Polarity: a special waveform used to test loudspeaker polarity
- Sine: a sine wave
- Pink Noise: a pseudo random noise signal whose frequency content falls as  $1/\text{frequency}$ .
- White Noise: a pseudo random noise signal whose frequency content is flat and has no frequency dependence
- Silence: a zero output waveform
- Sweep: a sine wave that changes frequency linearly between a low and a high limit
- Log Sweep: a sine wave whose frequency changes logarithmically between a low and a high point
- Warble: a sine wave whose frequency oscillates between a low and a high value







For the periodic signals, like Sine, Square etc., the frequency of the wave may be set using two methods:

- Via the text entry box (showing "800" in the example above) - simply enter the desired frequency, in Hertz. You must press the "Set Frequency" button after entering the desired frequency.
- Via the "<<", "<" etc. buttons - these buttons adjust the frequency in large and small steps.

To turn the generator output on and off, use the large button so labelled.

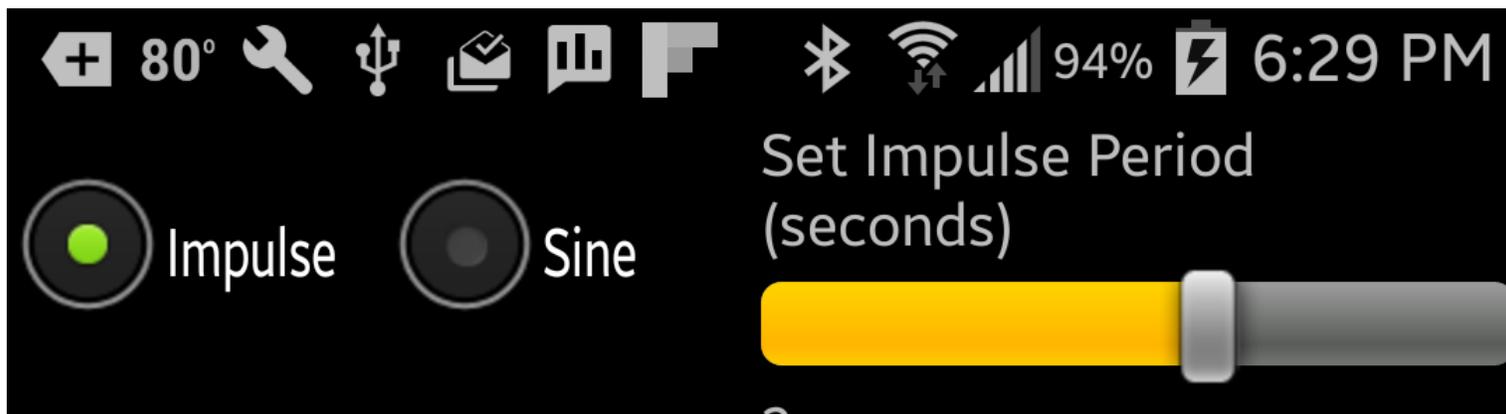
The volume of the generated signal can be adjusted using the slider - you can also use the hardware buttons on your device.

By default, the signal is sent to both channels, Left and Right. You can disable output on one or both of the channels using the checkboxes above the volume slider.

Technical note: The precision with which AudioTool can generate a given frequency depends on the internal sample rate being used and the sample buffer length. Typically, you can expect AudioTool to generate a frequency within 10Hz or less of the value you specify.

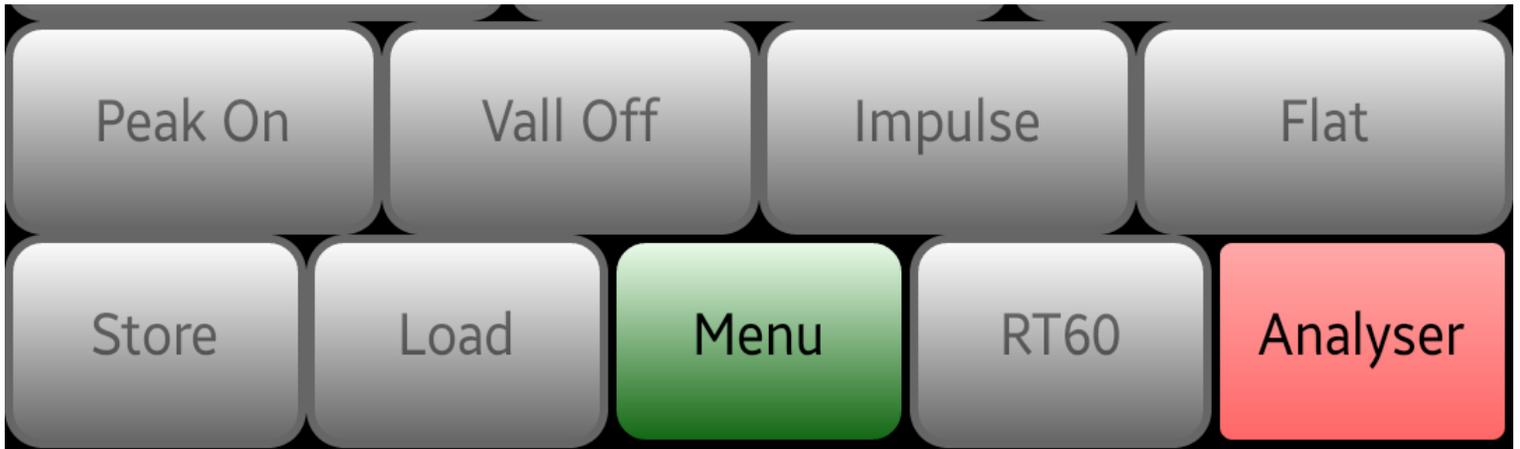
## Impulse

In those mode, periodic impulses are produced. You can set the interval between pulses using the slider control for that purpose.



The screenshot displays the AudioTool interface with the following elements:

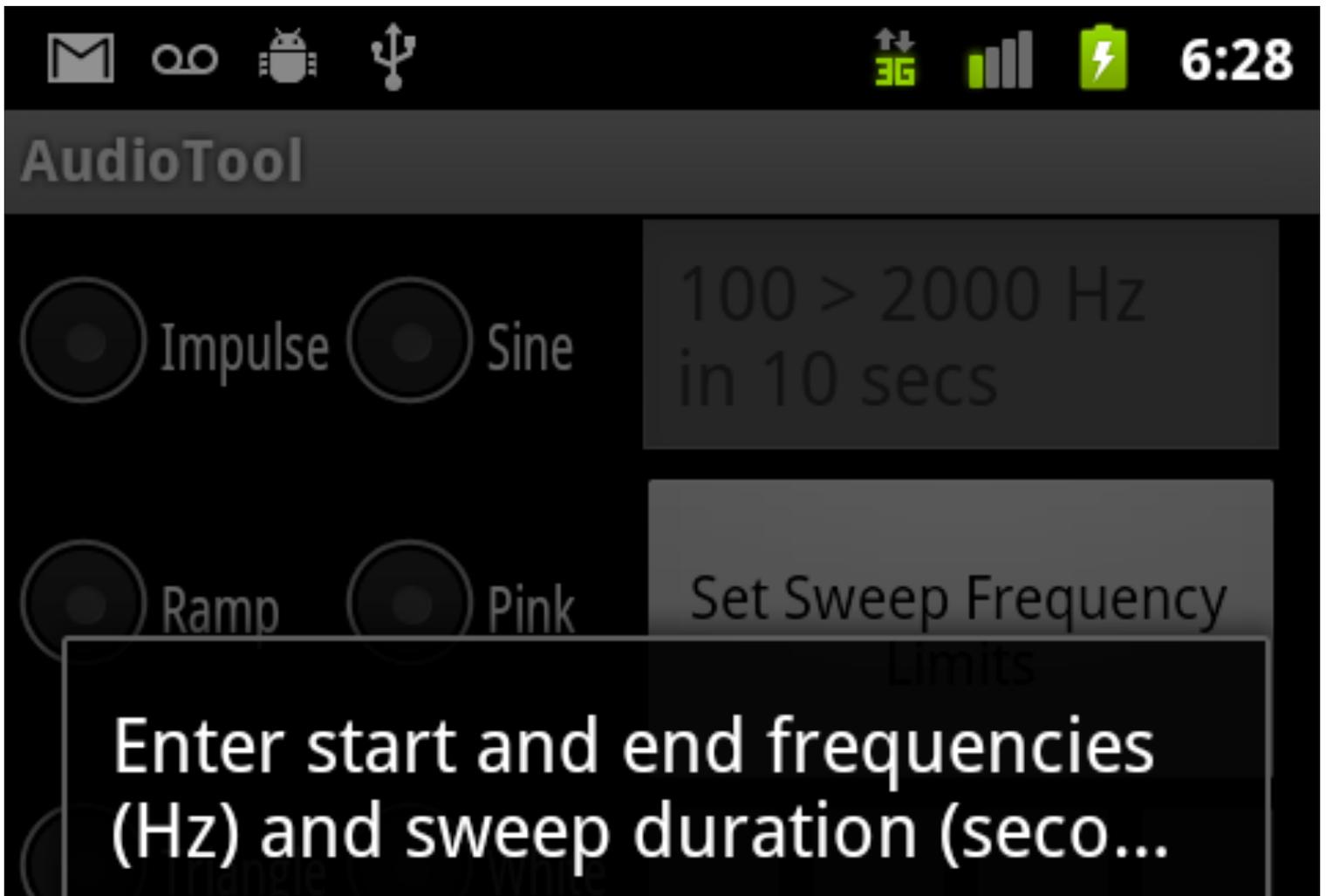
- At the top center, the text "3 SECS" is visible.
- A large grey button labeled "Off" is positioned in the upper right area.
- On the left side, there are ten radio button options arranged in two columns:
  - Row 1: Ramp, Pink
  - Row 2: Triangle, White
  - Row 3: Square, Silence
  - Row 4: Polarity, Sweep
  - Row 5: Warble, Log Sweep
- In the center-right, there are two checkboxes labeled "L" and "R", both of which are checked with a green checkmark.
- Below the checkboxes is a horizontal yellow slider with a grey knob on the right side.
- At the bottom, there are three large grey buttons labeled "Running", "Avg. Off", and "1/6 Octave".

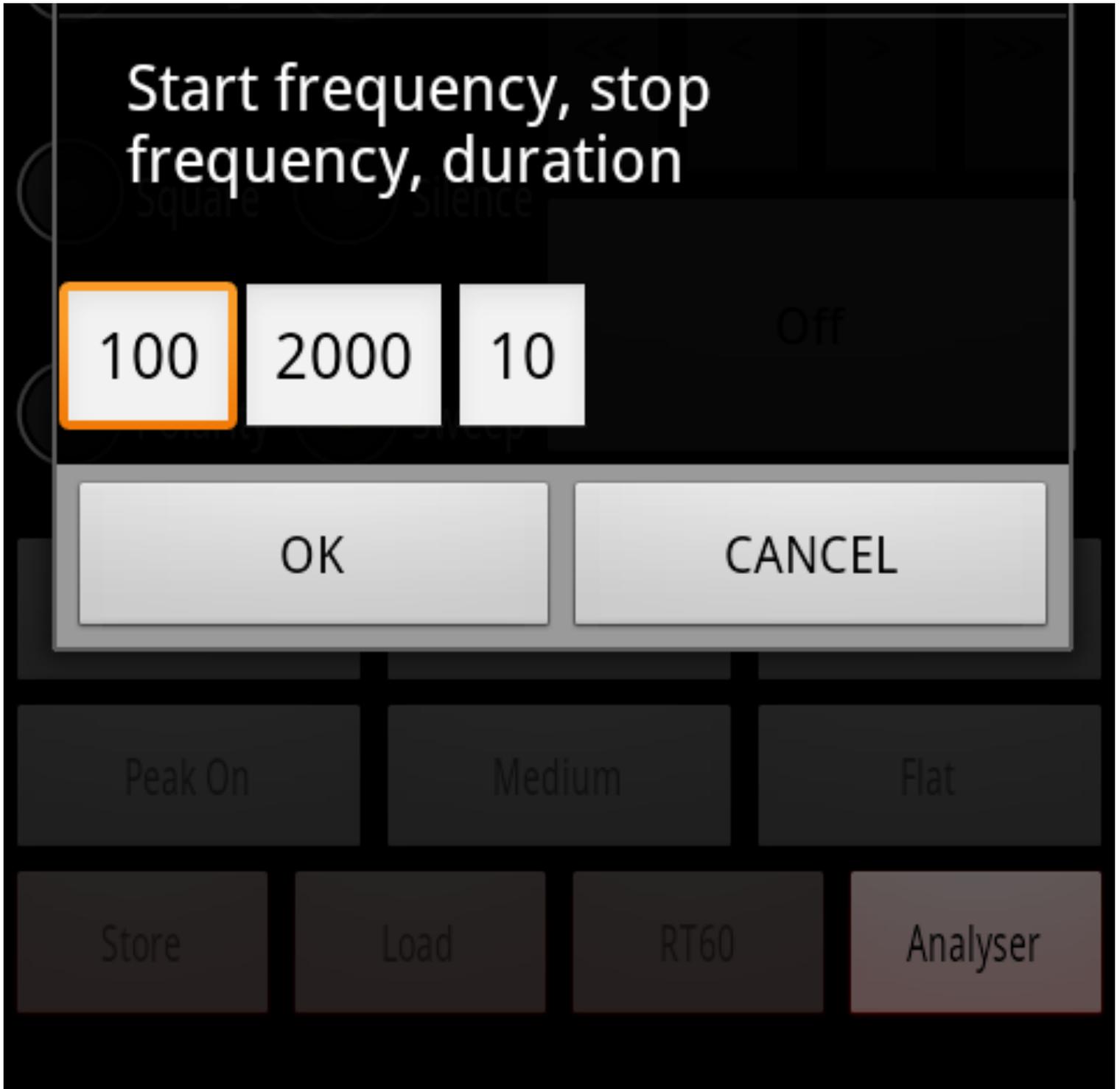


## Sweeps

When the Sweep or Log Sweep button is selected, AudioTool will generate swept sine waves.

To adjust the start and stop frequencies, and the duration of the sweep, use the "Set Sweep Frequency Limits" button to bring up the dialog box shown below:





In the example above, the settings will cause a sine wave starting at 100Hz to be swept up in frequency to 2000Hz, over a duration of 10 seconds.

## Warbles

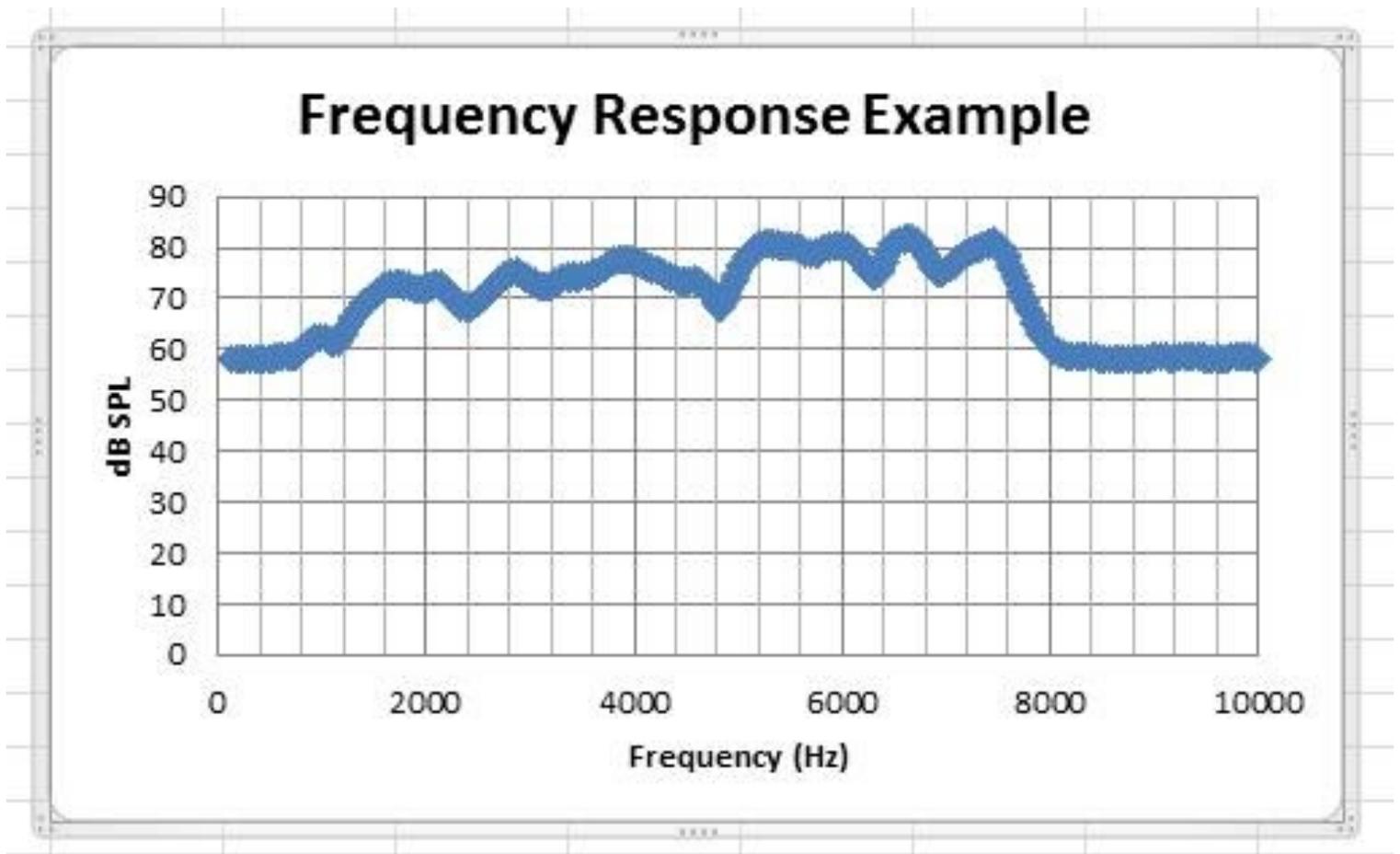
AudioTool will generate warbles when the Warble button is selected. You can adjust the low and high frequencies of the warble, and the duration between those limits. E.g if you set the low frequency to 400Hz, the high frequency to 800Hz, and the duration to 1.0 seconds, then you will hear a sine wave that starts at 400Hz, rises to 800Hz in half a second, then falls to 400Hz in half a second, and that cycle will repeat.

## Frequency Response with Sweeps

**Note that you can now more conveniently measure frequency response using the Peak Store mode described above (introduced in v7.0).**

If the generator is in Sweep mode, then the current sweep frequency is stored for every value of SPL measured by AudioTool. The current sweep frequency is also displayed on the main screen. In addition, if Chart Recorder mode is selected, and the chart saved using the Store button, then the SPL readings coupled with the generator frequency will be saved in the file.

The chart file may then be read in to e.g. Excel and a frequency response curve created by plotting the SPL readings versus the frequency values. Here is an example:



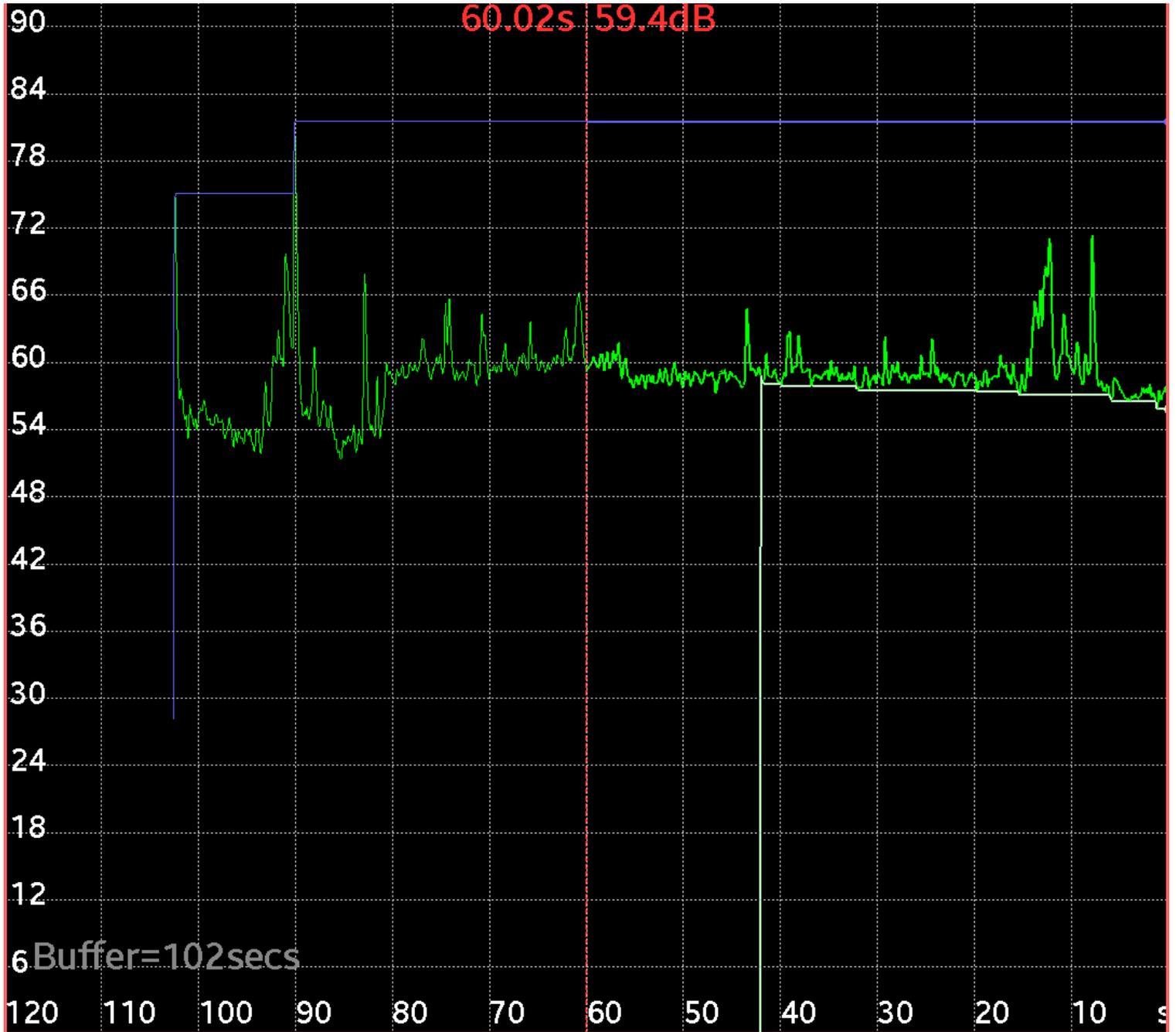
Each row of the Chart Recorder file contains the time, the measured dB SPL, the detected peak frequency, and optionally the sweep generator frequency at that time if sweeps were running at the time.

## Cursors

There are two cursors that can be enabled and disabled via the Menu, using the Show Cursors or Hide Cursors options.

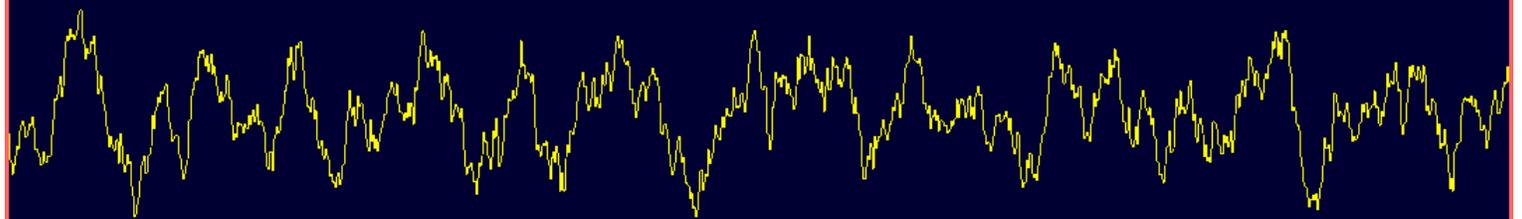
Initially, both cursors are positioned mid-screen - you can move the cursors simply by touching them and sliding your finger to the desired position. In RTA mode, the cursors show the frequency and dB level at the cursor position. In Chart Recorder mode they show the time and dB SPL:

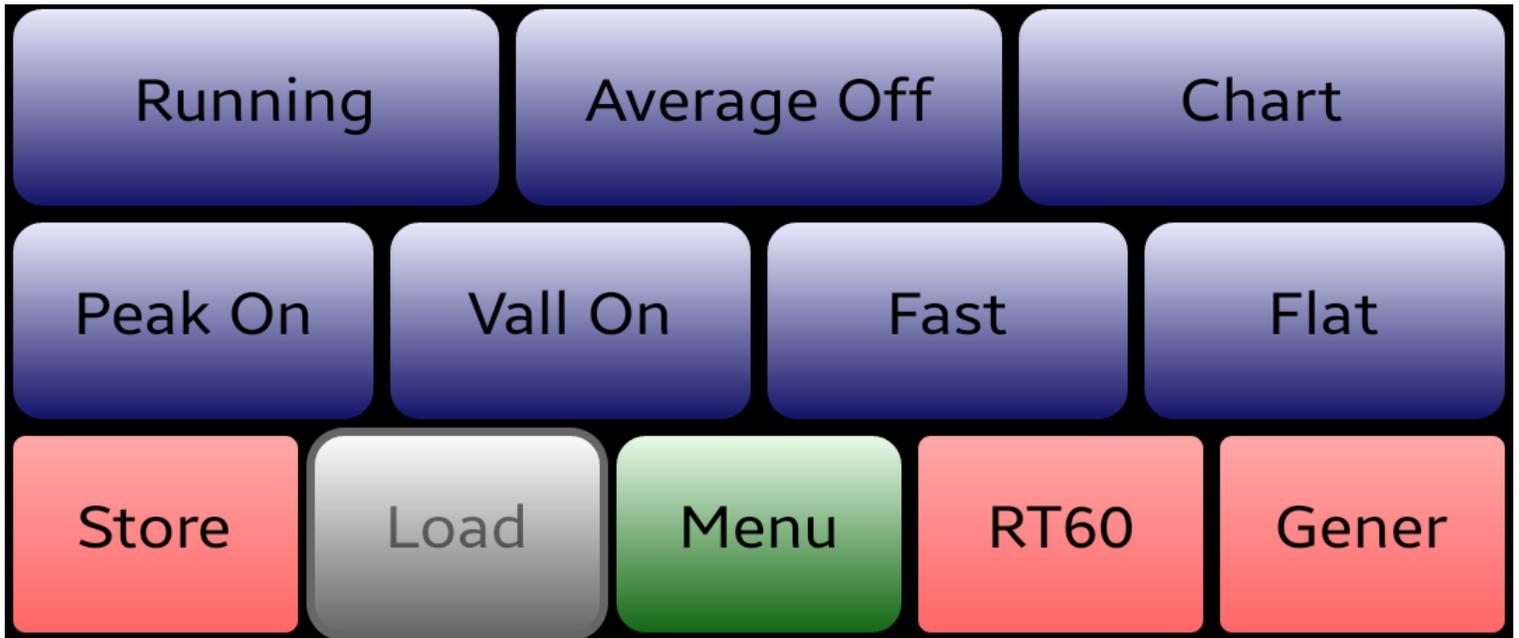




Mic 44100 11 8192 dF=5Hz C=1/3oct O=5.0

57 dB SPL





## Sone Loudness Measure

New in v7.5! Sones are a measurement of the perceived loudness of sound, which takes into account the sensitivity of the human ear to different frequencies.

An SPL level of 40dB at 1kHz sounds much louder than an SPL level of 40dB at, say, 125Hz. The Sone level of the 40dB sound at 1kHz is about 1.0, whereas the Sone level of the 40dB sound at 125Hz is about 0.1 ... ten time softer. More information about Sones may be found at the Wikipedia entry:

<https://en.wikipedia.org/wiki/Sone>

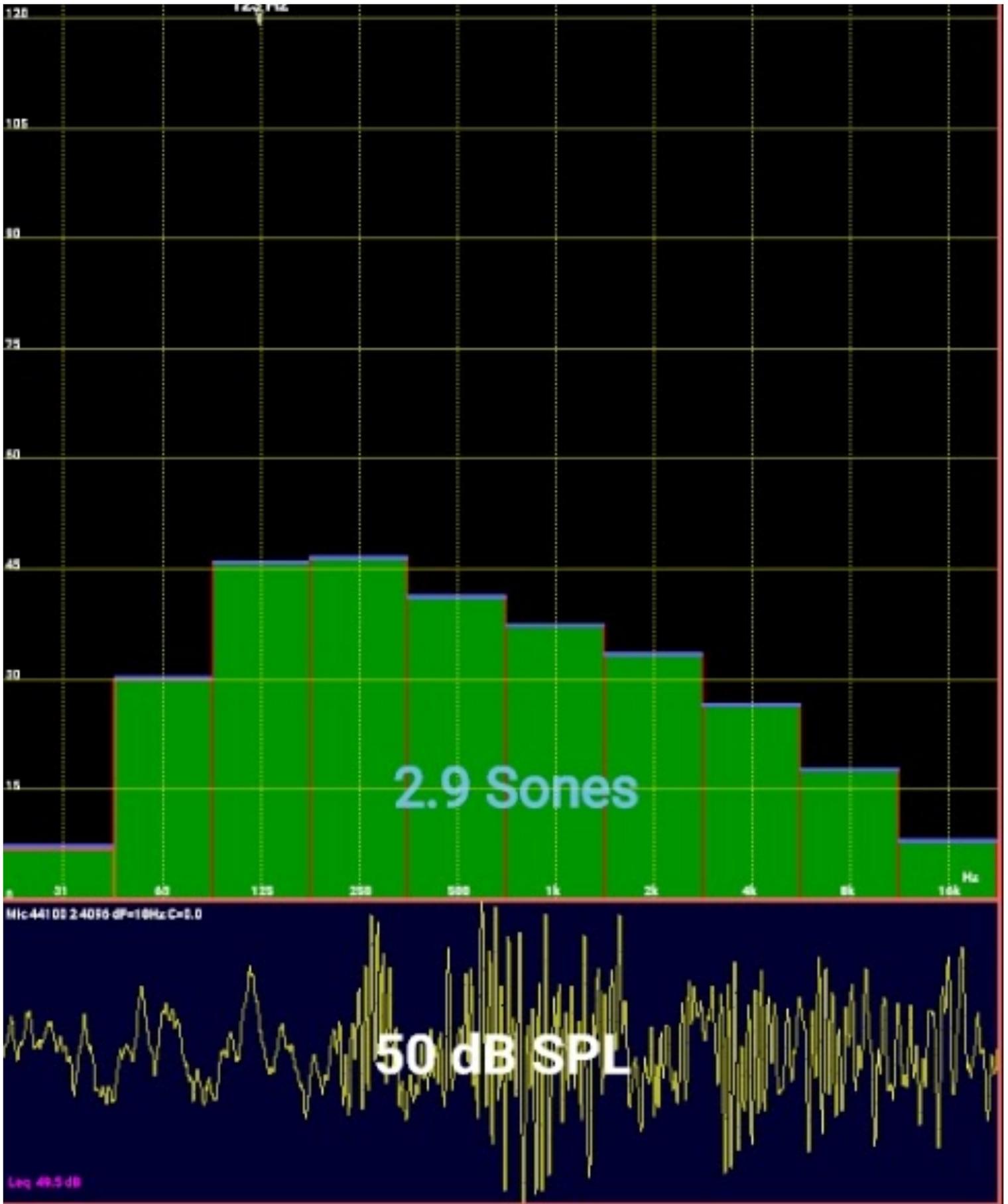
AudioTool calculates an estimated Sone level based on real time 1/1 octave SPL measurements. The SPL levels in the 63, 125, 250, 500, 1k, 2k, 4k and 8k bands are used together with the HVI Equal Loudness Index values (See HVI Publication 915: [https://www.hvi.org/ratings/HVI915\\_091013.pdf](https://www.hvi.org/ratings/HVI915_091013.pdf)) to calculate a Sone level using the formula:

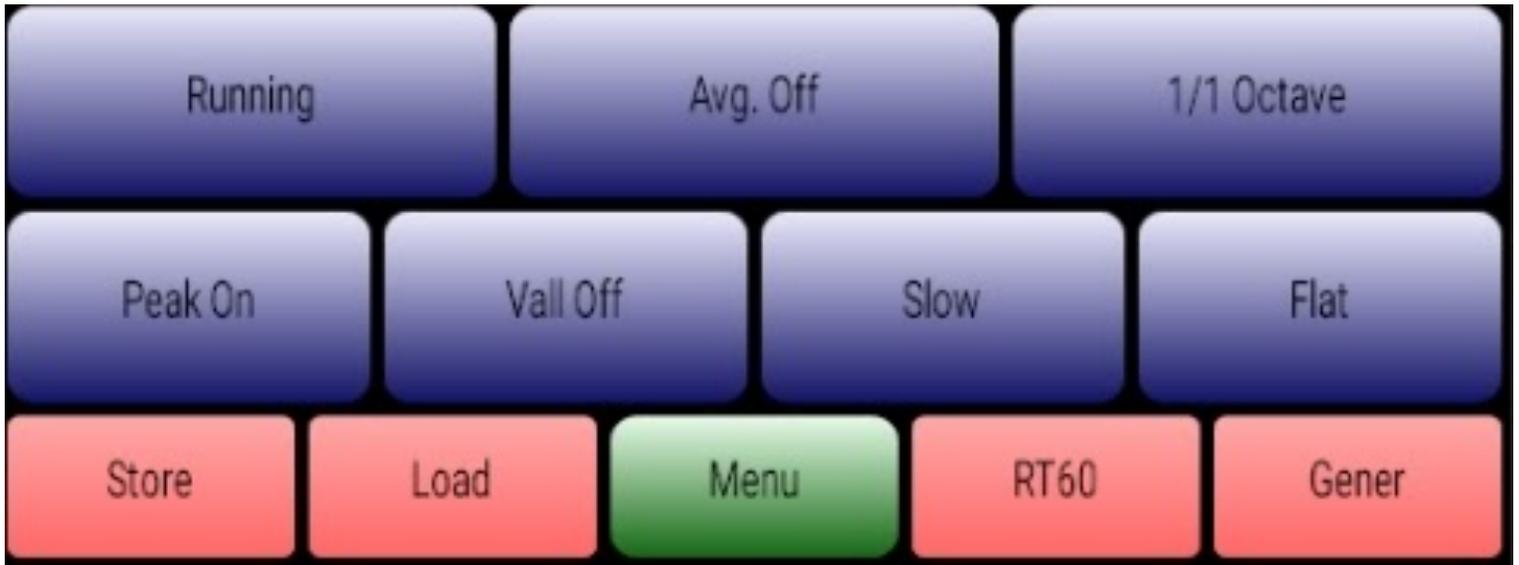
$$\text{Sone} = 0.7 * (\text{Loudness @ 1kHz}) + 0.3 * (\text{Sum of Loudnesses, all bands})$$

To display the Sone measure, use the Menu option "Show Sone Measure". (You should set AudioTool to "Flat" weighting, otherwise the Sone measure will be underestimated.)

In 1/1 Octave mode you will see the calculated Sone value:







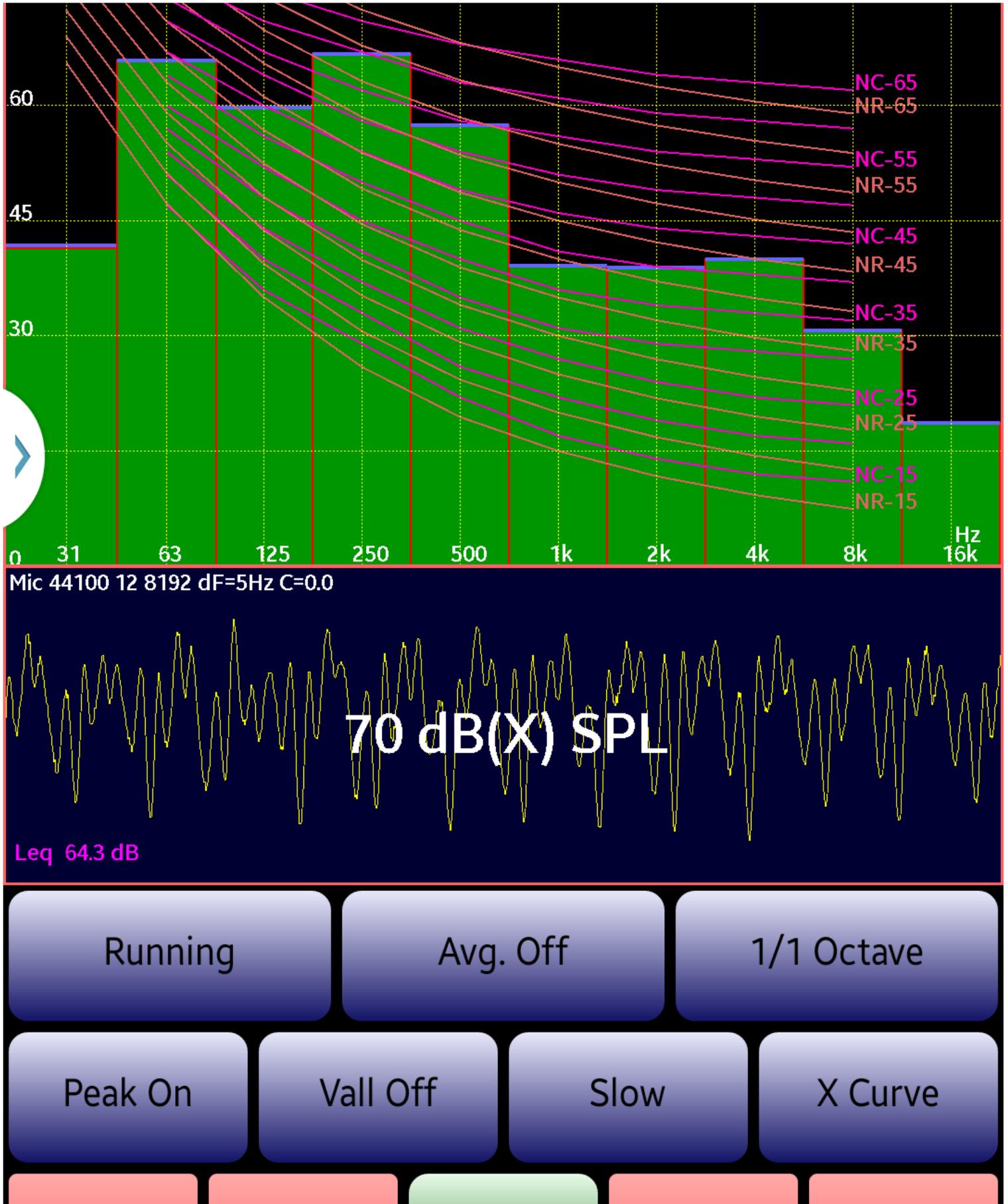
## Noise Curves (NC and NR)

In 1/1 Octave mode you can display the Noise Curve (NC) family, as well as the NR family (for European users) via the main menu, and AudioTool will calculate the prevailing NC and NR values.

For more information on NC Noise Curves see [here](#).

For more information on NR Noise Curves see [here](#).





Store

Load

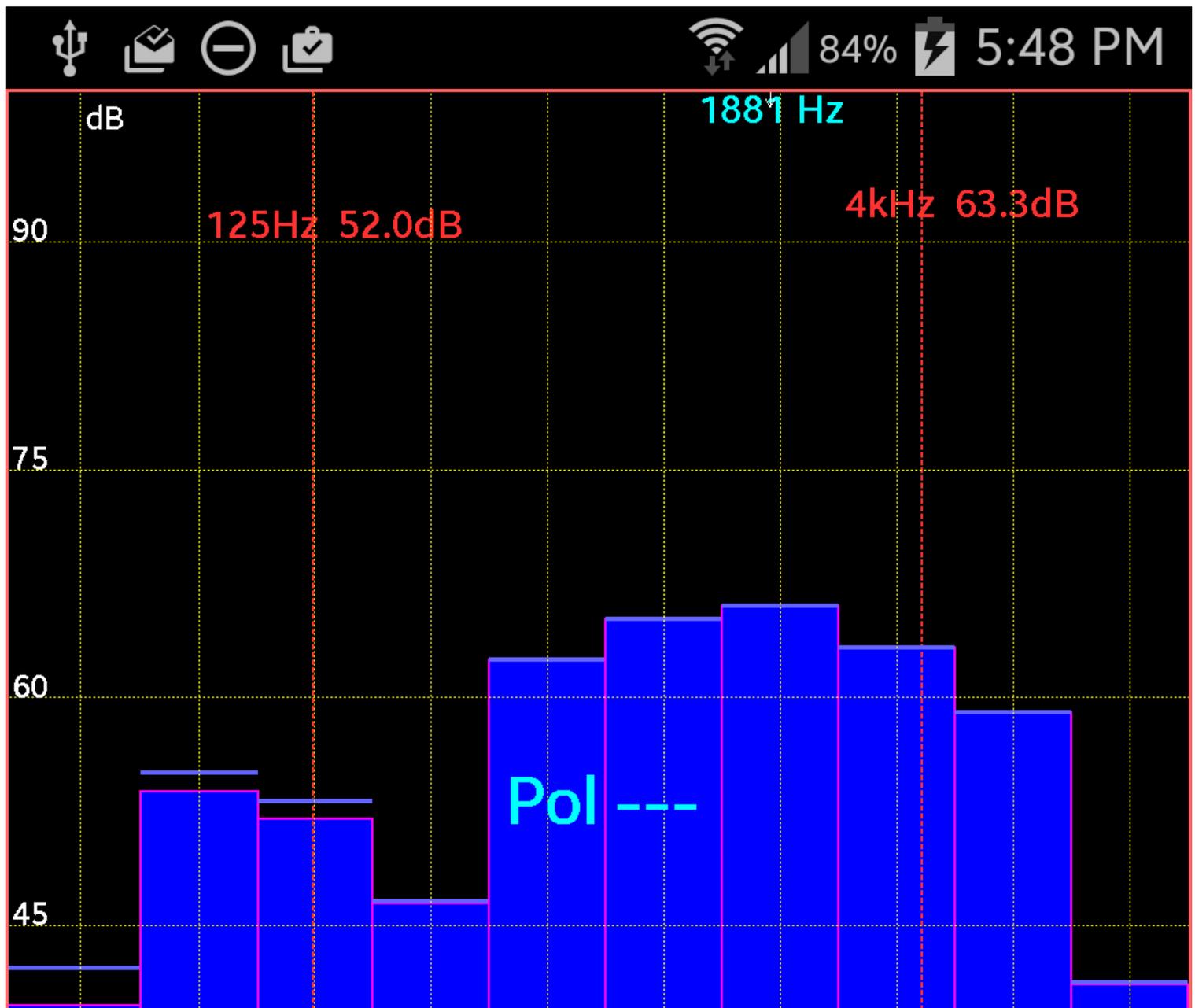
Menu

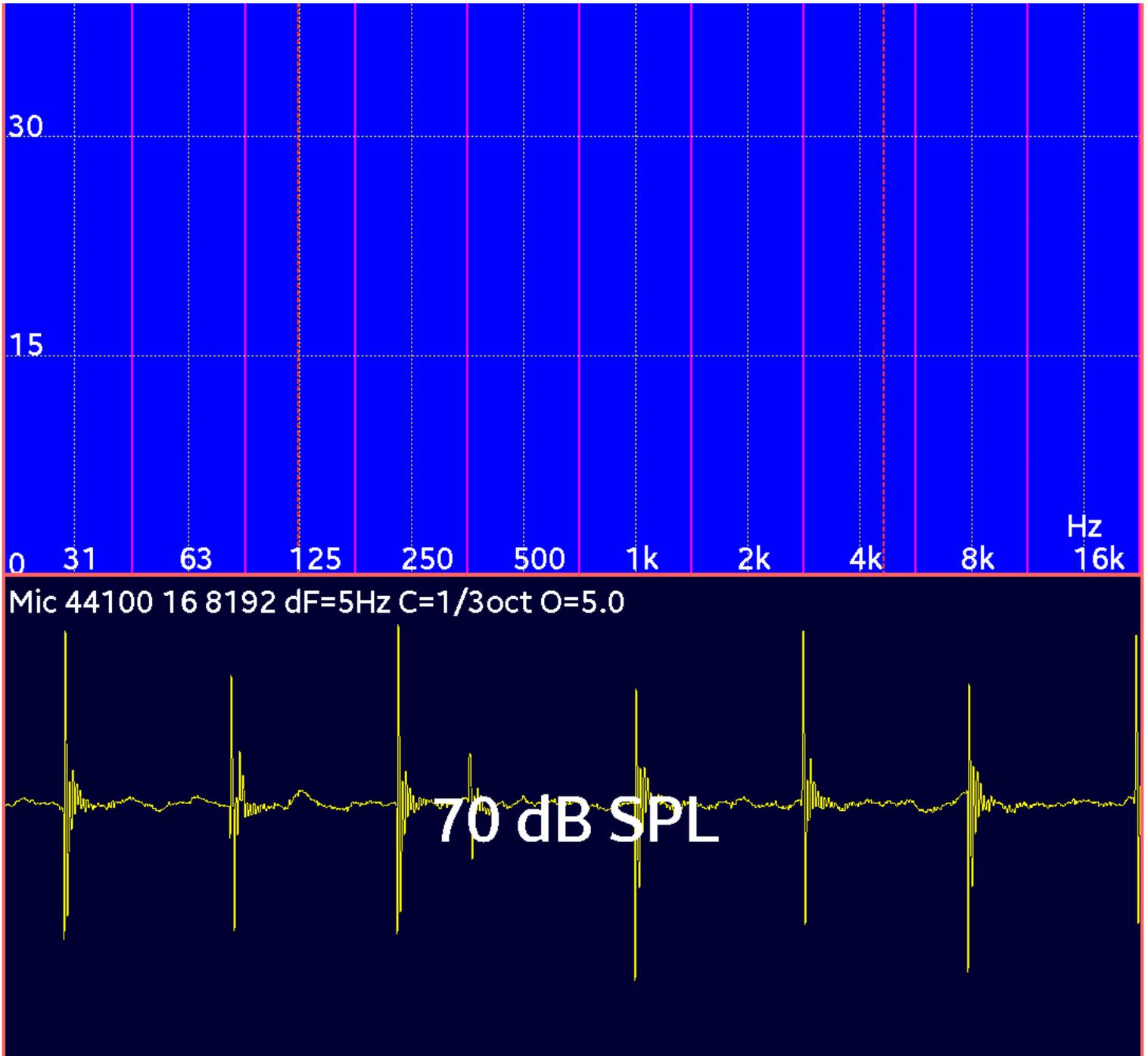
RT60

Gener

## Polarity Checking

The loudspeaker polarity checker can be used to check whether a loudspeaker is wired in phase: connect the AudioTool output to the speaker under test, select "Polarity" from the signal generator screen, then return to the RTA screen. If the speaker is out of phase (polarity reversed) AudioTool will show "Pol ---", otherwise "Pol +++" if the speaker is in phase.



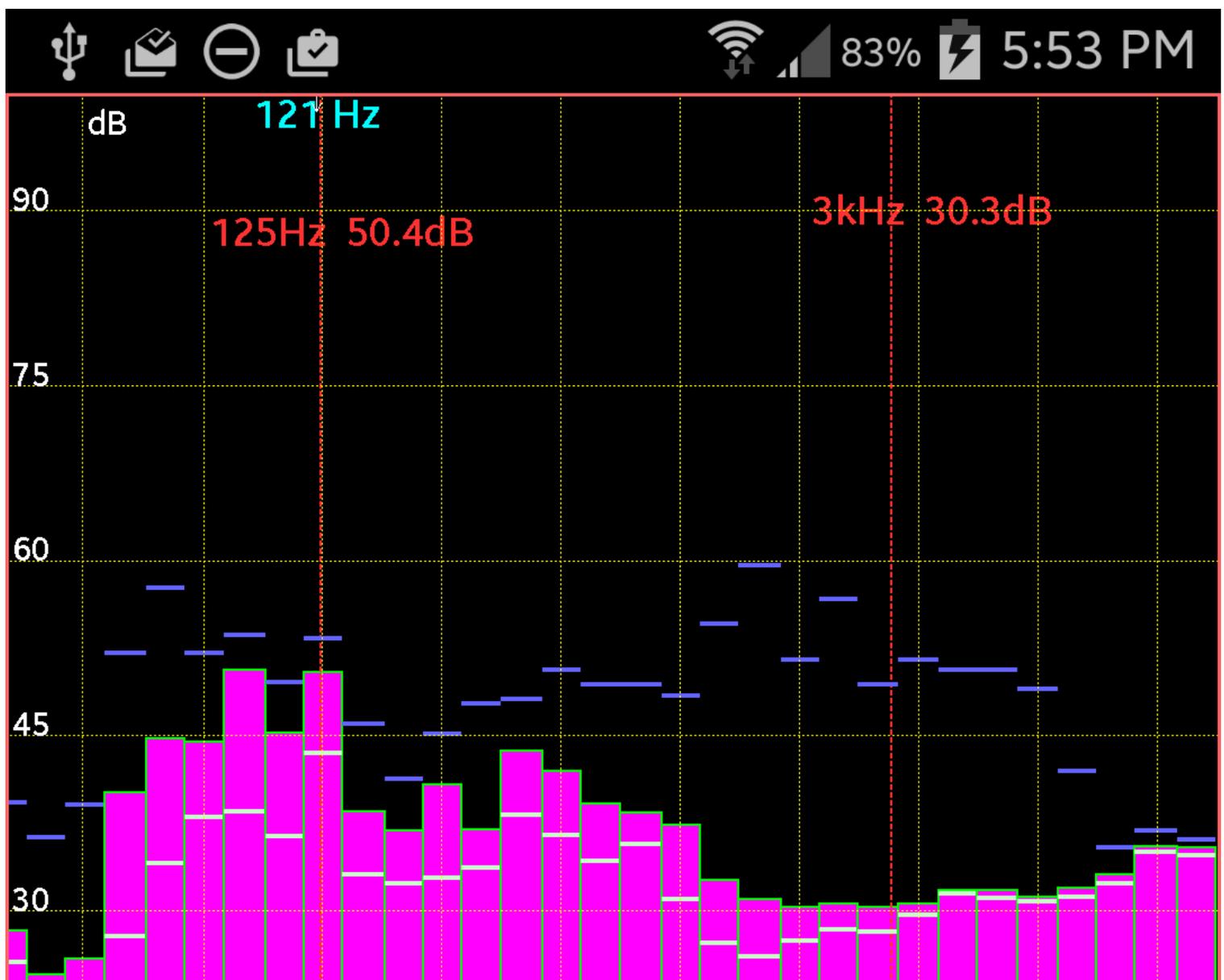


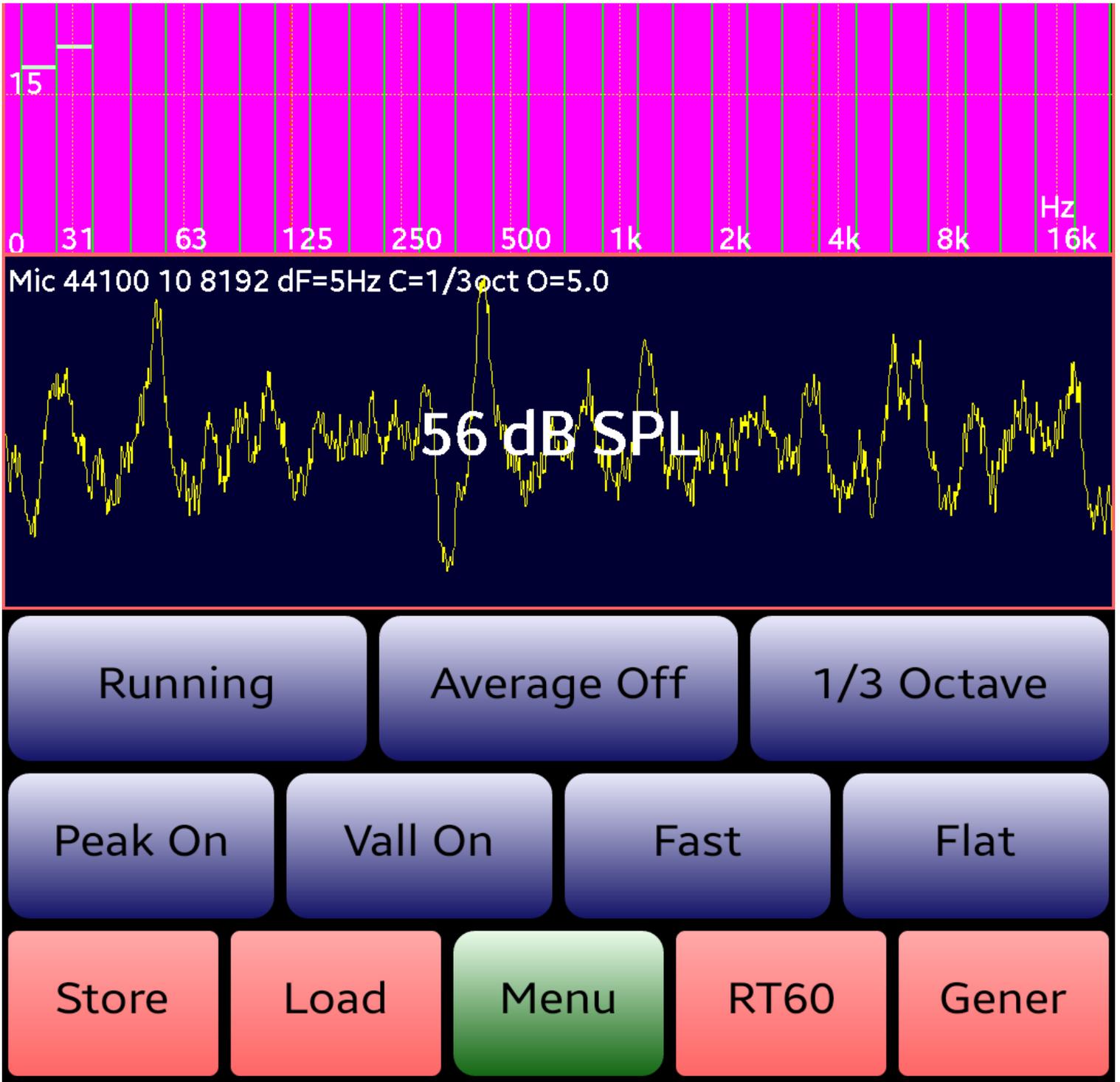
## Peak Detect

With Peak On, AudioTool keeps track of the maximum dB levels observed at each frequency, and shows these as blue bars (or a blue line) on the screen. To reset the peak levels, simply turn Peak off then on again.

## Valley Detect

With Valley On, AudioTool keeps track of the minimum dB levels observed at each frequency, and shows these as white bars (or a white line) on the screen. To reset the valley levels, simply turn Valley off then on again. Here is a screenshot showing both peak and valley enabled.





## Filter Speeds

Four speeds are provided, selected by the "Medium"/"Fast"/"Impulse"/"Slow" button in the RTA screen. This control selects the speed of the RTA bars and the dB SPL measurement. Exponential filters are used with time constants of 0.5, 0.125, 0.025 and 1.0 seconds, respectively.

## Weighting

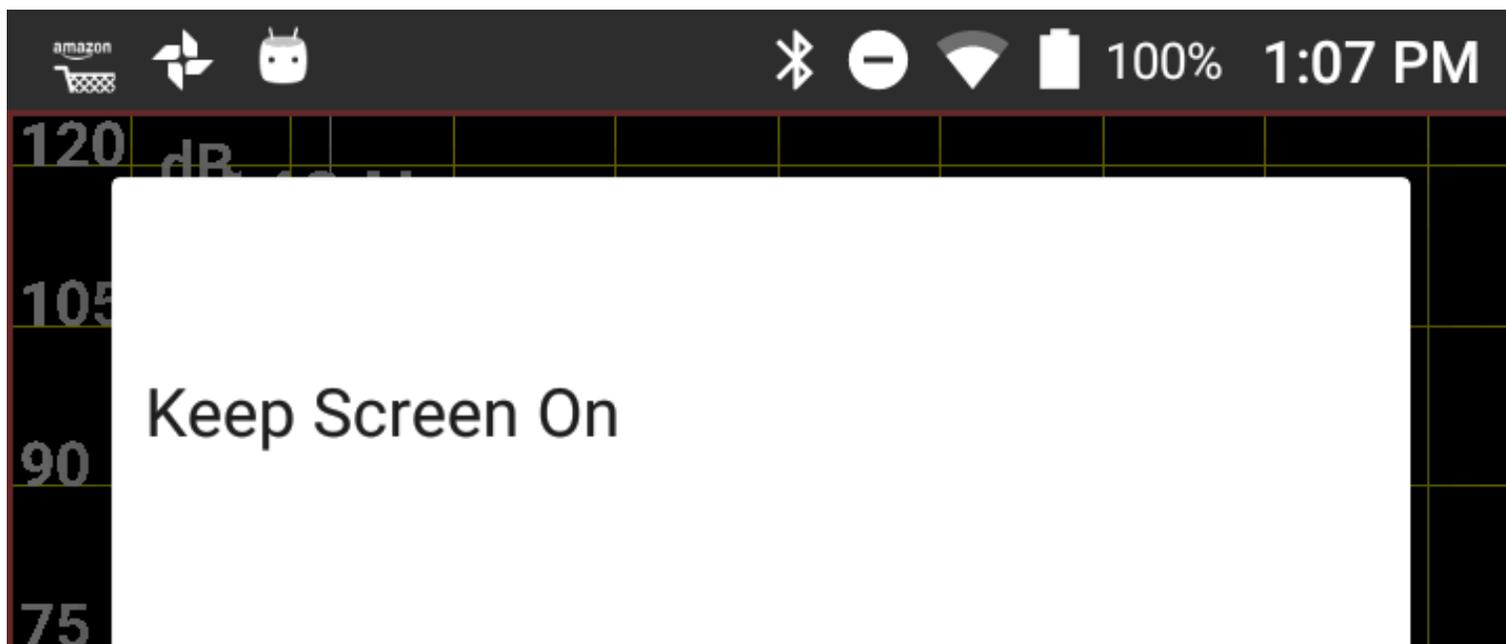
Four weights are provided: no filter (flat), A and C weight, and X Curve filters, which are applied to the raw audio data and thus affect the RTA levels and the dB SPL level. The X Curve weighting is implemented as a 2kHz Low Pass filter and a 60Hz High Pass filter in series: the frequency response is flat and rolls off 3dB/octave above 2kHz and below 60Hz.

## Averaging

Frequency response data may be averaged - if averaging is enabled then the dB levels at each frequency are averaged for each sample that is acquired.

## Screen Control

You can prevent the screen rotating when you rotate your Android phone or tablet by using the "Prevent Screen Rotate" options in the Menu:



# Prevent Screen Rotate

Help (web)

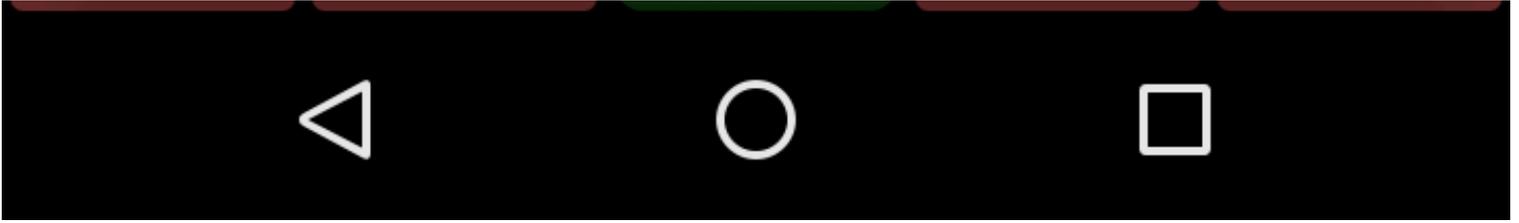
Discussion Group (web)

Version Info

Please Rate AudioTool

Exit

CANCEL



Some users prefer to have the screen locked in this way. To revert to the normal behaviour use the "Allow Screen Rotate" Menu option.

In addition, the screen can be prevented from "sleeping" by using the "Keep Screen On" Menu option, and allowed by the "Allow Sleep" option.

## Font Size

You can adjust the font size used for the text in the AudioTool screens and displays via the Menu. There are four options: Small Font, Large Font, Massive Font and Ginormous Font. Changing the font cycles through these in order.