The Psychoacoustic Bass Enhancer

by Jan Meier

[Editor: The author has applied for a patent in Germany for the invention described in this article. The German Patent Application number is 10119094.8. Individuals are authorized to make this device for their own personal use, but must obtain permission from Jan Meier for commercial applications.]

Our senses, especially our eyes and ears, are remarkably precise instruments. We can distinguish the slightest gradations in intensity, color, frequency, etc. Nonetheless, eyes and ears are also easily fooled. For instance, in well known optical experiments straight lines may look bent or equally long lines seem to have different lengths These experiments tell us, that the brain processes involved in perception also play an important role in the way we see and hear our world.

The study of the physiological and mental processes of hearing is called psychoacoustics. Principles of psychoacoustics are widely used in audio technology. An example of psychoacoustic processing is data compression with MPEG-3, which removes information from the signal without (or almost without) affecting sound quality. Another example is the loudness button on many amplifiers that compensates for the reduced sensitivity of the ear for the highest and lowest frequencies at low sound pressures.

In this article a device is presented that makes signals below 60 Hz audible in loudspeakers and headphones that normally, by their mere physical construction, are not able to reproduce these frequencies. The device combines two psychoacoustic phenomena.

BACKGROUND

The principle of the missing fundamental:

The sound of a single note of a music instrument is the summation of its fundamental tone (say 200 Hz) and a number of harmonics (400, 600, 800, 1000, ... Hz). If we electronically remove the 200 Hz fundamental tone our ear only hears the harmonic frequencies at 400 Hz and higher. Nonetheless our brain tells us that the pitch of the note is 200 Hz. Since the 600 and 1000 Hz frequency components are no fundamentals of the lowest (400 Hz) frequency component present, our brain knows that something is missing and “adds” an imaginary fundamental tone of 200 Hz. However, the “color” of the note is lighter when the fundamental tone is missing.

Mechanical harmonic distortion in the inner ear:

A pure 200 Hz sine wave not only makes the basillary membrane inside the ear vibrate at 200 Hz, but also at 400, 600, 800, 1000, 1200.... Hz. For sine waves between 200 and 3000 Hz these overtones have amplitudes of 33%, 13%, 6%, 4%, 2%, ... of the amplitude of the fundamental. Harmonic distortion in the inner-ear thus sums up to approximately 60%!

Nonetheless, we only hear a pure sine-wave at the fundamental frequency since our brain has learned that this specific frequency spectrum belongs to a pure tone.

To my knowledge, nobody has ever investigated the frequency spectrum of the basillary membrane at very low frequencies. However, the decreasing sensitivity of the ear indicates that the membrane is relatively "stiff" for this frequency range. Therefore it can be expected that the relative amount of overtones at the lowest frequencies strongly increases. This phenomenon is seen with many music instruments. A "cheap" piano does not really produce a 27.5 Hz tone when one strikes the lowest key, because the resonance board is simply too small to swing at 27.5 Hz, but it produces a tone that appears to be 27.5Hz due to the principle of the missing fundamental.

At their lowest frequencies corpses are more ready to vibrate at the overtones than at the fundamental. My guess was that in the inner ear the information on the lowest tones thus will be merely transmitted by the overtones produced.

Most loudspeakers and headphones are not able to make the air move at frequencies between 20 and 50 Hz and therefore these frequencies will not be heard. However, if we electronically create harmonics of these lowest tones and add these signals to the original audio-signal, we suddenly will hear the low fundamentals, due to the principal of the missing fundamental. Moreover, my speculation was, that if the spectrum of these overtones was chosen so as to create an energy spectrum on the basillary membrane that, except for the fundamental tone, resembles that of a pure sine wave, then we will hear something that is very close to this sine tone. This idea is illustrated in figure 1.

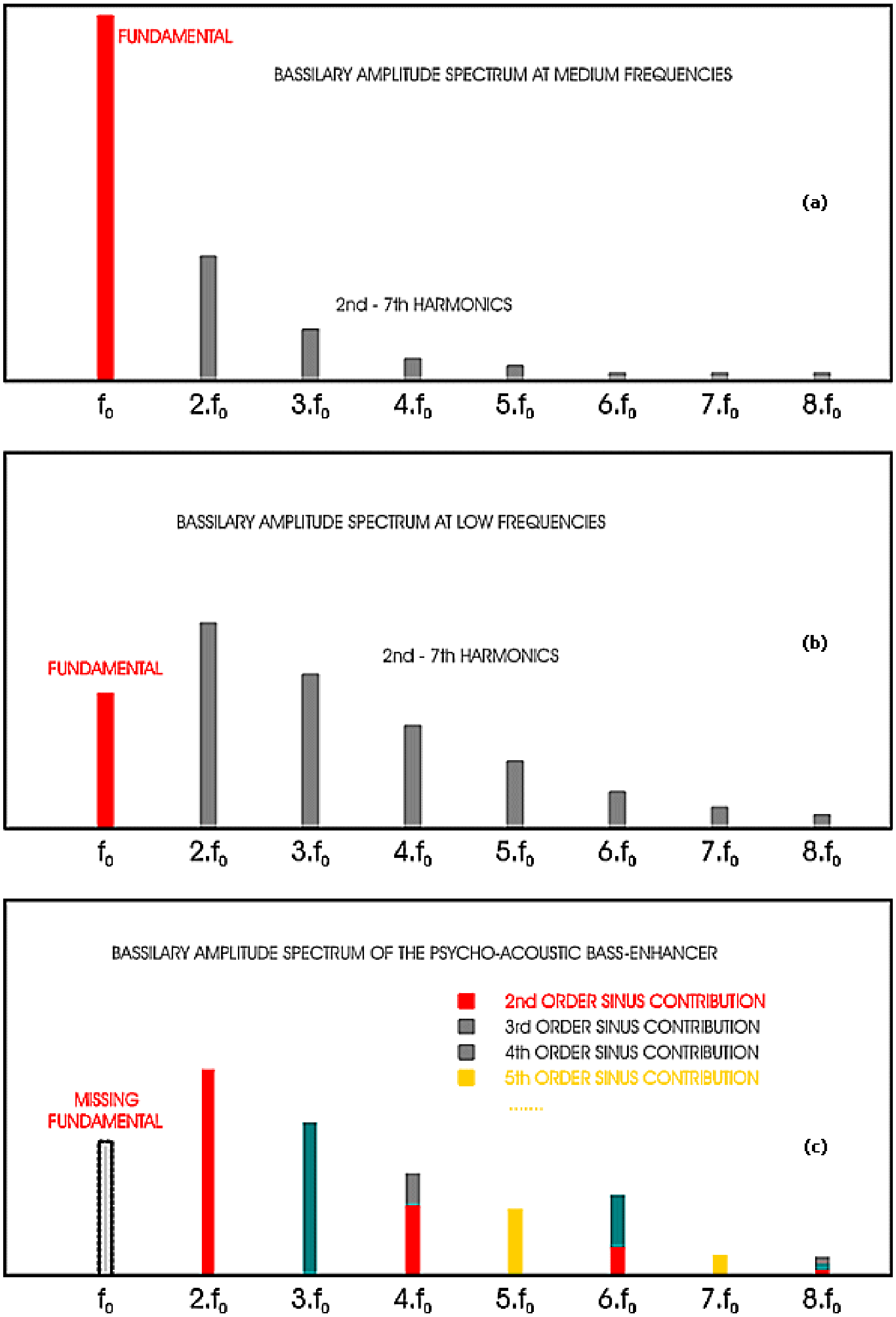


Figure 1

The most accurate way to generate a spectrum of overtones is to have the audio signal analyzed and frequency components added by using the technique of Fourier analysis. However, this requires quite a lot of computational power and off-line evaluation before the music can be played. For a more practical use, I wanted an analogue, real-time solution.

One analogue solution to create overtones already exists. The Philips company uses it for portable equipment (boomboxes and the like) and calls it Ultrabass. Basically this solution looks at the low frequency content of the signal and restarts a continuously increasing triangular signal at each zero-crossing of the signal.

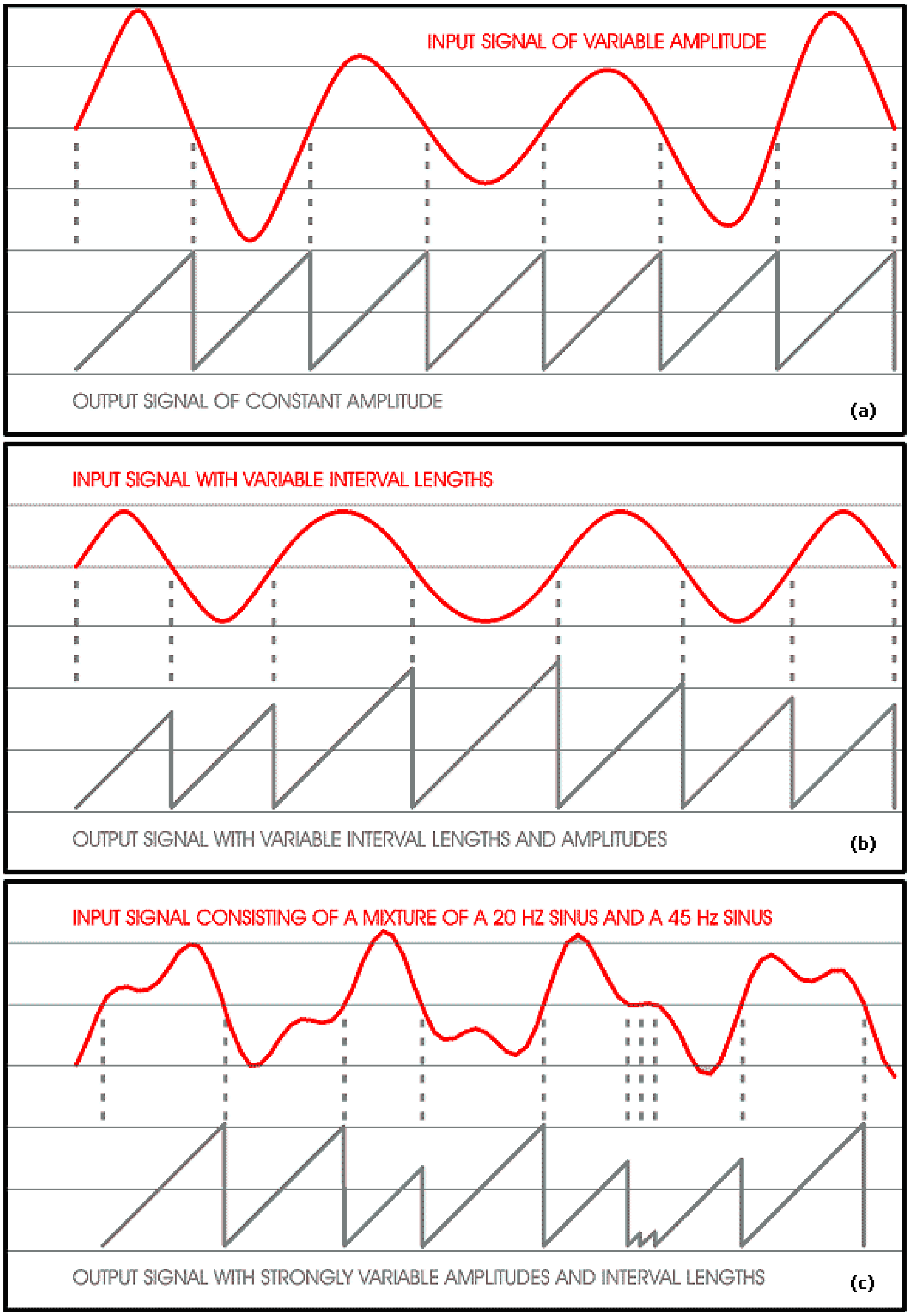


Figure 2

The resulting triangular waveform has a fundamental frequency twice that of the original waveform (see figure 2) and is added to the original signal. Ultrabass has a number of disadvantages that makes it hard to call it HiFi:

- With a sinusoidal input signal, only 2^{nd}, 4^{th}, 6^{th}, ... (even order) harmonics are created, because UltraBass generates even and odd harmonics from a sawtooth waveform, whose basic frequency is twice the frequency of the original input signal.

  So f(1) = 2 x f(0)

  The harmonics are even harmonics of the original input signal

  f(2) = 2 x f(1) = 4 x f(0)

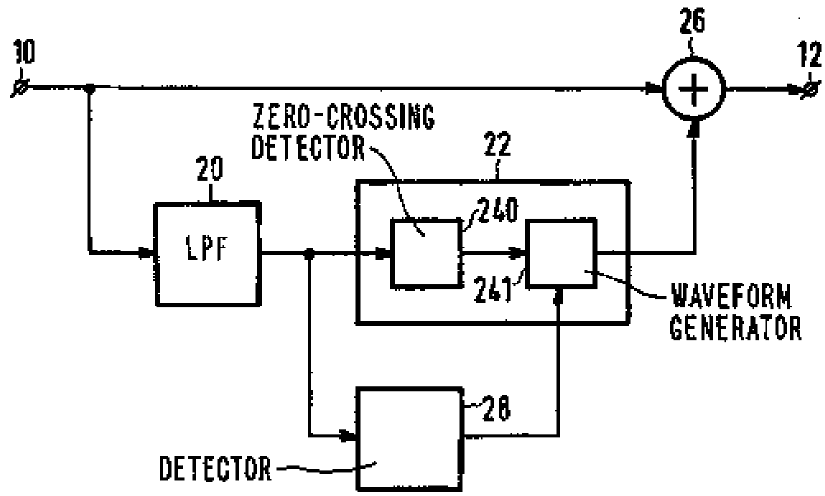
  f(3) = 3 x f(1) = 6 x f(0)

  f(4) = 4 x f(1) = 8 x f(0)

f(5) = 5 x f(1) = 10 x f(0)
.....

However, to "hear" the missing fundamental we also need the 3rd, 5th, ... (odd order) harmonics.

- Short term variations (cycle-to-cycle) in signal amplitude are not reproduced in the Ultrabass signal, because the amplitude of the Ultrabass signal is set by its envelope and only is allowed to vary slowly. These variations are very important to our ears for recognizing signals as being from a non-artificial origin. (Figure 2a)
- Short term variations in cycle length are reflected in the Ultrabass signal but the longer cycles also have larger amplitudes. (Figure 2b) This is unnatural. Short term variations are very important to recognize an instrument as not being artificial. However, adding short term variations that are not there in reality will change our perception of the instrument.
- With a mixture of two signals of different frequencies the Ultrabass signal merely represents the strongest signal. It is the strongest signal that sets the number of zero crossings in the Ultrabass harmonics generator. However, the presence of the second signal results in a number of aberrations. One is shown in figure 2c.



UltraBass Circuit Diagram (src: US Patent No. 6111960)

Figure 3

Those interested in more details on Ultrabass can take a look at the US patent no. 6111960 ("Circuit, Audio System and Method for Processing Signals, and a Harmonics Generator" by Aarts et al.).

I decided to design my own solution that had to fulfill the following requirements:

- all lower harmonics (2nd, 3rd, 4th, 5th, 6th, 7th, ...) should be calculated.
- The amplitude of the harmonics should decrease with increasing order. The rate of decrease should be adjustable (I did not have any indication yet, what would be the most suitable ratios of the various amplitudes so making it adjustable allowed me to experiment).
- Cycle-to-cycle variations in signal amplitude and cycle length should be properly reproduced in the calculated harmonics.
- With a mix of two signals, harmonics of both fundamentals should be calculated.

Such a design is not very straightforward. I have tried several solutions - none being ideal - but the one presented in this article is close to optimal and still relatively easy to build. The basic schematics are shown in Fig 4.

THE CIRCUIT

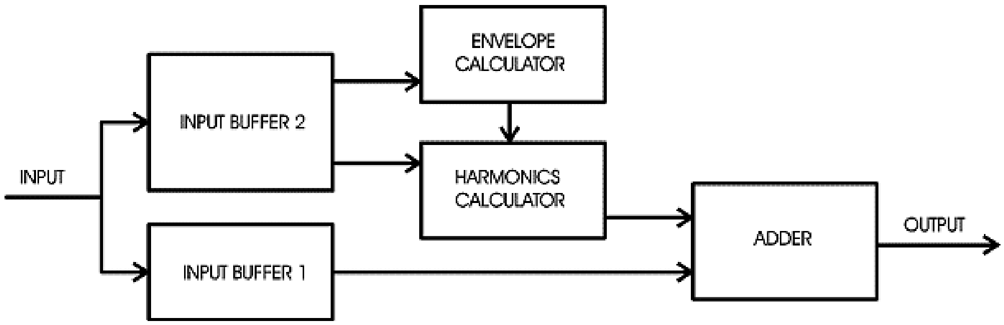


Figure 4

The input signal, $V_{in}(t)$, is buffered by two parallel buffer stages. Input buffer 1 leaves the input signal unaltered. Input buffer 2 filters the signal and removes the high frequency (and the very low frequency) components.

A harmonics generator takes the filtered buffer signal, $V'_{in}(t)$, and uses it as input for a high order mathematical function.

$$V_{gen}(t) = a_2.V_{in}^2(t) + a_3.V_{in}^3(t) + a_4.V_{in}^4(t) + \ldots\ldots\ldots$$

It is a multiplication series of the form: f(out) = a.f(in) + b.f(in).f(in) + c.f(in).f(in).f(in) + ...

If the filtered input signal is a pure sine wave, this procedure generates all the major harmonics. For example the square of a sinusoidal signal generates a signal with the double frequency (plus an offset):

$$\sin^2(2p.f.t) = 0.5\ (1 - \cos(4p.f.t))$$

If the amplitude of a signal decreases by a factor two, then the squared signal would decrease by a factor four! Thus, weak signals that pass through the generator would not be heard and strong signals would be “amplified” too strongly. To correct for this phenomenon a separate circuit continuously monitors the signal envelope and adds this information to the harmonics generator. The generator corrects the amplitude of its output signal accordingly.

The last stage of the device is an adder to sum the original input signal, the generator signal, and (for offset correction reasons) the envelope signal.

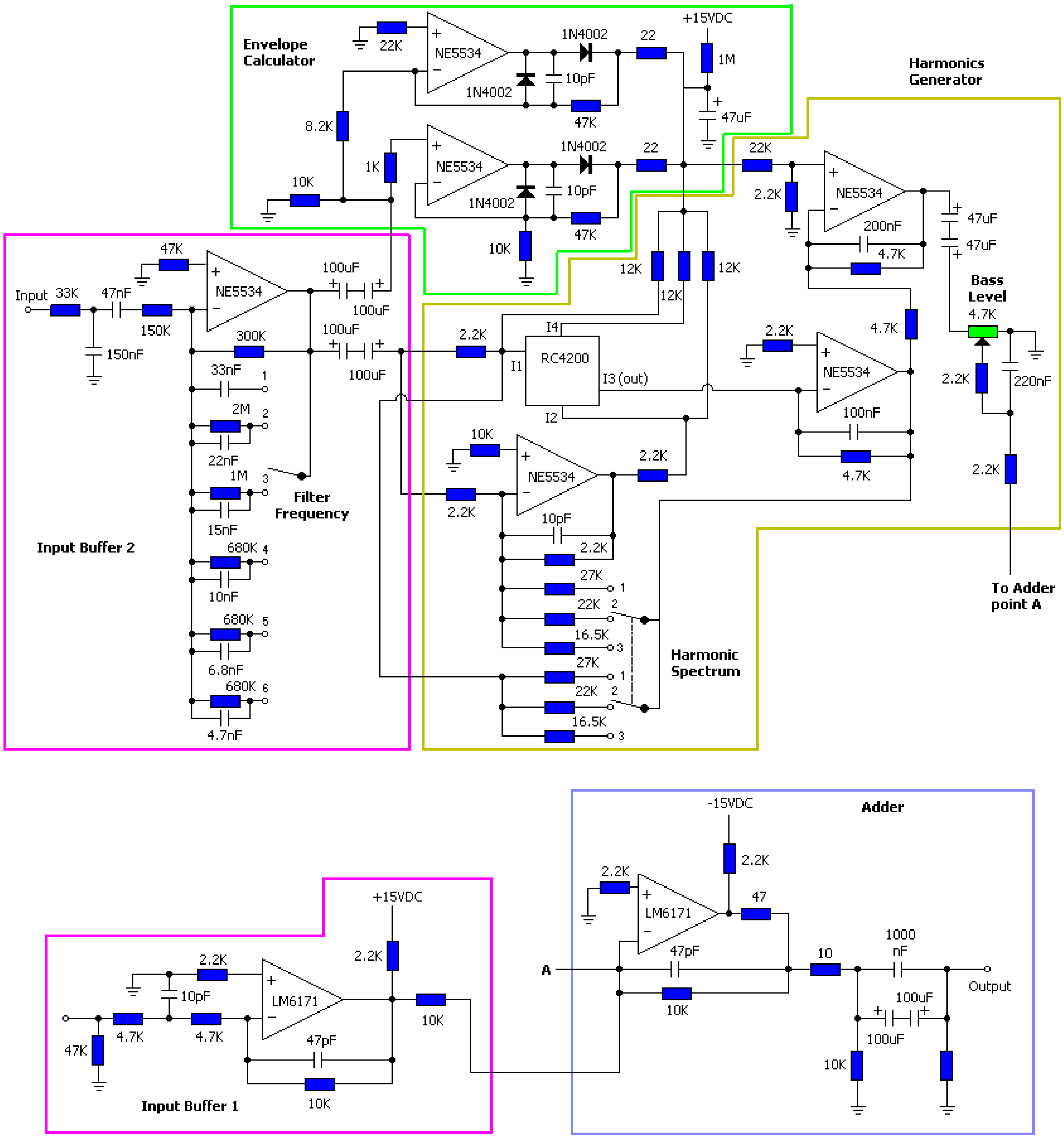


Figure 5

Figure 5 shows the schematics in detail.

Input buffer 1:

This input buffer is inverting. Since the adder is also inverting, the total signal path is non-inverting

Input buffer 2:

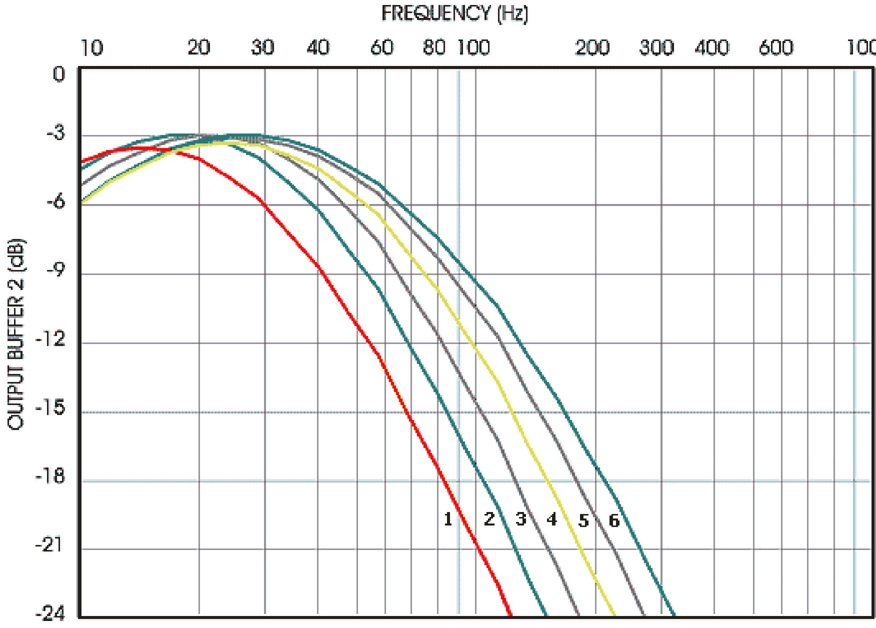


Figure 6

This input stage combines a 1st order high-pass filter with a second order low-pass filter. The filter characteristics can be changed by adjusting the resistance and the capacitance inside the feedback loop. Figure 6 shows the filter characteristics for the various settings.

The envelope calculator:

The envelope signal, A(t), should increase instantaneously at a sudden increase of the amplitude of the input signal and slowly decrease after a single wave. This is achieved by having two opamps charging a 47uF capacitor as soon as the amplitude of the input signal is higher than the voltage across this capacitor. One opamps charges at the large positive signal amplitudes, the other inverts the signal and charges at the large negative signal amplitudes. Two 22 Ohm resistors limit the envelope signal A(t) at the capacitor to frequencies up to 150 Hz. The capacitor slowly discharges via three 12 kOhm resistors that connect the envelope calculator and the harmonics generator.

The harmonics generator:

Central to the harmonics generator is an analogue current multiplier/divider RC4200. This IC has three inputs I_1, I_2, I_4 that are actively kept at ground potential and one output I_3. The input and output currents are allowed positive values only and are related according to:

$I_3 = (I_1 \,.\, I_2) / I_4$

Basically, with the harmonics generator the input signals are given by

$I_1 = A(t)/12000 + V'_{in}(t)/2200$

$I_2 = A(t)/12000 - V'_{in}(t)/2200$ (the filtered buffer signal has passed a converter)

$I_4 = A(t)/12000$

For a sinusoidal input signal the envelope signal is approximately 5.7 times (amplification factor of the envelope converter) the sine amplitude.

$V'_{in}(t) = K.\sin(2p.f.t)$

$A(t) \sim 5.7\ K$

$I_3 \sim (K/2100 + K.\sin(2p.f.t)/2200)(K/2100 - K.\sin(2p.f.t)/2200)/(K/2100) = K/2100\ (1 - 0.91.\sin^2(2p.f.t))$

The output signal of the RC4200 thus is a cosine signal with twice the frequency of the input sine wave plus an offset. The amplitude of the cosine and the offset are proportional to the amplitude of the input signal.

The ratio of the amplitudes of the envelope signal and the input signals has been taken with care to create an offset in the output signal that is slightly higher than the time varying signal part. Thus it is made sure that all currents are positive.

The envelope generator is connected via a 1M ohm resistor to the 12 Volts power supply. This guarantees that I_1, I_2 and I_4 have small positive values, even when the input signal is zero. If the input signals are zero the RC4200 becomes instable.

The output signal of the RC4200 could well be used as input signal for another RC4200 to multiply it with the original sinus signal. Thus signal components with three times the frequency of the input signal can be calculated. However, to continue this way to construct the higher order harmonics is quite cumbersome. Moreover, the RC4200 is an expensive part, so using a single RC4200 for each higher order harmonics would become quite expensive. I therefore used a trick and connected the output signal of the RC4200 (after buffering with a opamp) to its own two inputs by means of variable resistors R_f. Thus a feedback is created that generates the higher harmonics. The feedback resistors R_f can be varied to create different harmonic spectra (a higher R_f produces more harmonics).

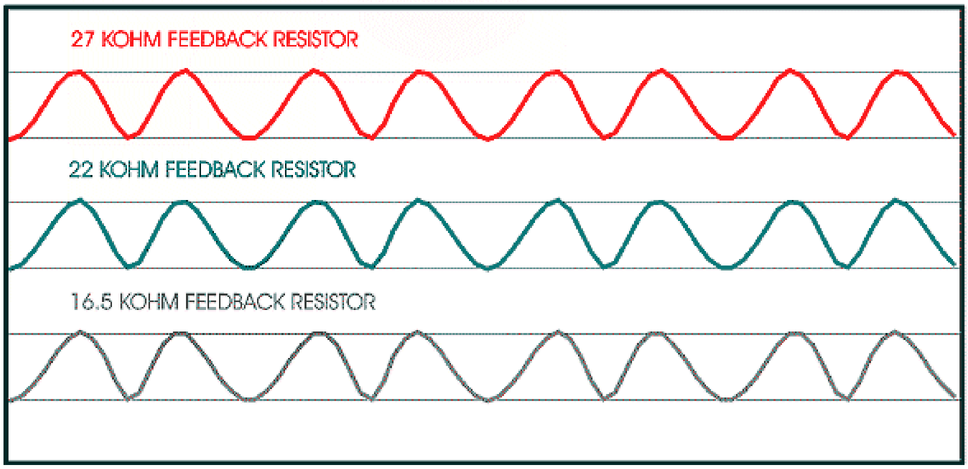


Figure 7a

Figure 7a shows the output signals and their amplitude spectra for a sinusoidal input signal with the three resistor value settings as shown in the schematics. They are very similar (figure 7b shows the three outputs from 7a close to each other for comparison). The differences are more easy to discern sonically and in the frequency spectra.

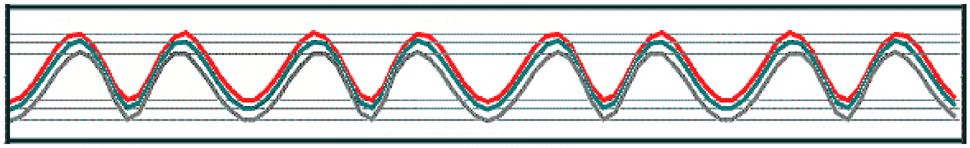


Figure 7b

The higher harmonics created are not expected to contribute strongly to the impression of deep bass tones. At the contrary, having many overtones could make the sound rather "sharp". Therefore the buffer at the output of the RC4200 constitutes a low-pass filter with an upper frequency of 340 Hz.

With no signal at the input the output signal of the RC4200 is very low. The sudden onset of an input signal instantaneously increases the output signal to a specific mean value. This signal jump is corrected by subtracting (half of) the envelope signal from the output of the RC4200. The subtractor for this purpose also constitutes a low-pass filter with an upper frequency of 340 Hz.

The output of the harmonics generator is added to the original input signal via a (linear) potentiometer that allows a continuous control of its amplitude.

CONSTRUCTION

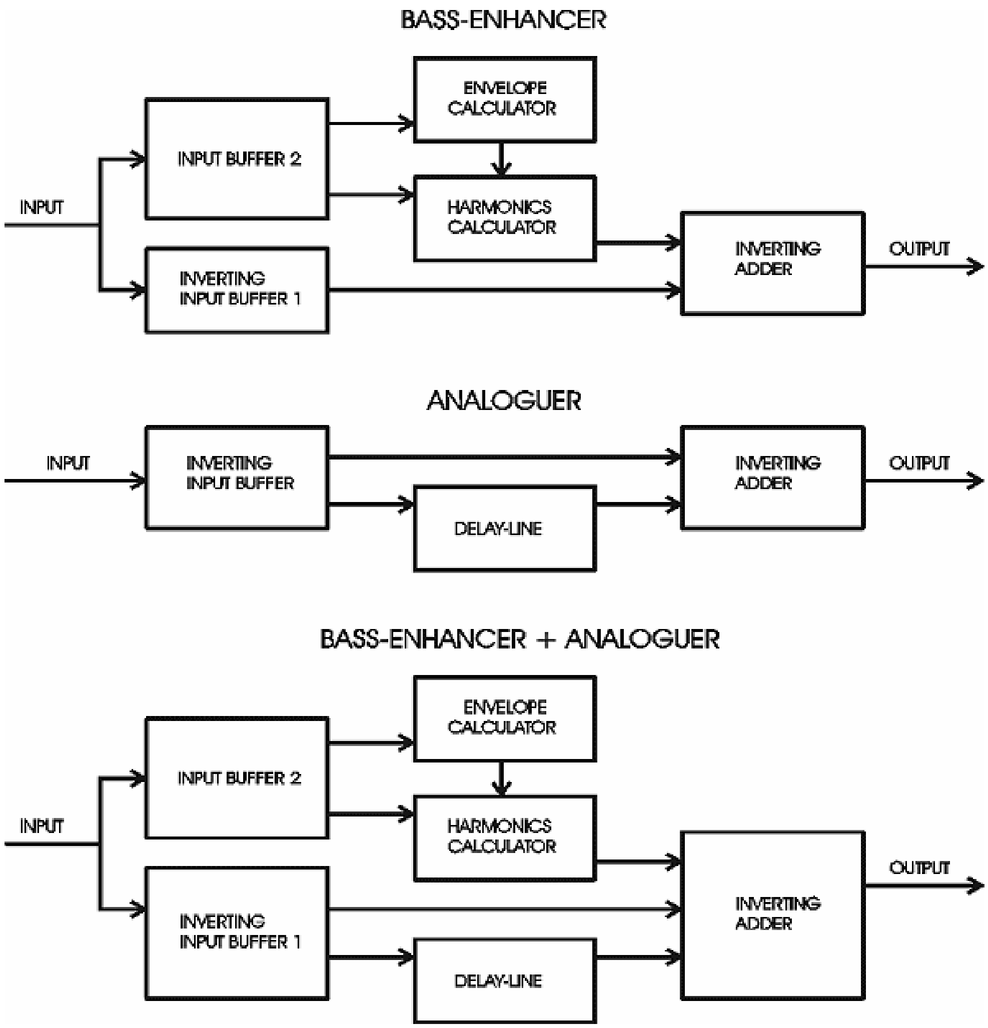


Figure 8

People that have read my article on the Analoguer will note a similarity. The Analoguer too has an input buffer at the front and an adder at its end. This allows for an easy integration of the two systems into one device (figure 8). Somebody who already has build the Analoguer only has to add the second buffer 2, the envelope calculator, the harmonics generator and the potentiometer.

For the direct signal buffer and the adder high quality opamps are recommended. I use LM6171 forced into class-A. For the other blocks sound quality is of less importance, since our signal is artificial anyway. I used relatively cheap dual N5534 opamps, not driven into class-A.

The RC4200 is a multiplier made by Fairchild and by JRC and costs about $7 US. There are more analogue multipliers on the market but these are generally much more expensive. There are no pin-to-pin compatible substitutes, however. I used an IC socket for the RC4200, but this is simply because it was a first version that was expected to need quite some "debugging". It allowed me to remove the chips to prevent frying them when soldering other components.

The rotary switches are made by Lorlin. I used metal-film 0.2 W no-name versions that I have on stock in abundance and some decent polyester capacitors. The potentiometer is a no-name type and should be linear taper.

In the picture, I have two enclosures that were put on top on each other. The Analoguer circuitry and the bass-enhancer circuitry were connected by cables. The bass-enhancer was built on experimental circuit board only and looks rather messy. The lower unit is a normal Analoguer. The upper unit is the bass-enhancer. The big dial on the right sets the volume of the bass signal added (potentiometer). The left small dial sets the filter frequency of the input buffer 2. The dial at the back of the unit sets the feedback resistance in the loop of the multiplier that sets the shape of the spectrum.

Due to the larger number of ICs the current demand is rather high. The power supply should be ±15 Volts, 200 mA minimum. I used two Analoguer power supplies (one power supply for the Analoguer and one for the bass enhancer branch), but people also simply could take a larger transformer. One Analoguer power supply will power both the Analoguer and the bass-enhancer if the transformer power handling is increased to 15VA.

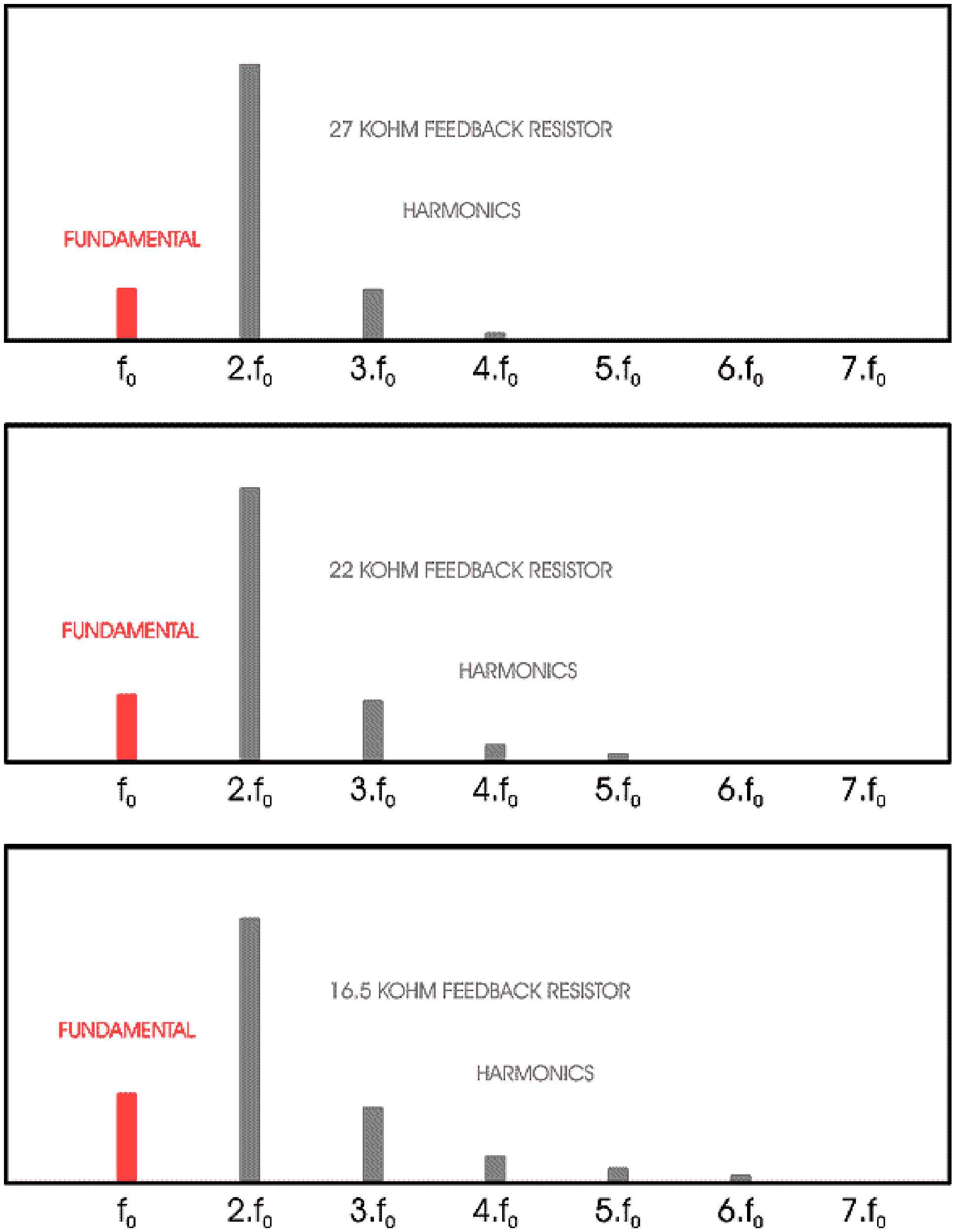


Figure 9

To check the proper function of the envelope calculator and the harmonics generator not only requires the use of at least an oscilloscope and a frequency generator (or CD-player with test-tones), but also a thorough understanding of the function of the various components. I do not have any specific test procedure. Figure 9 shows an example of the output signal for various sittings of the harmonic composition switch. The output of the harmonics generator can be measured at the 4.7K ohm potentiometer.

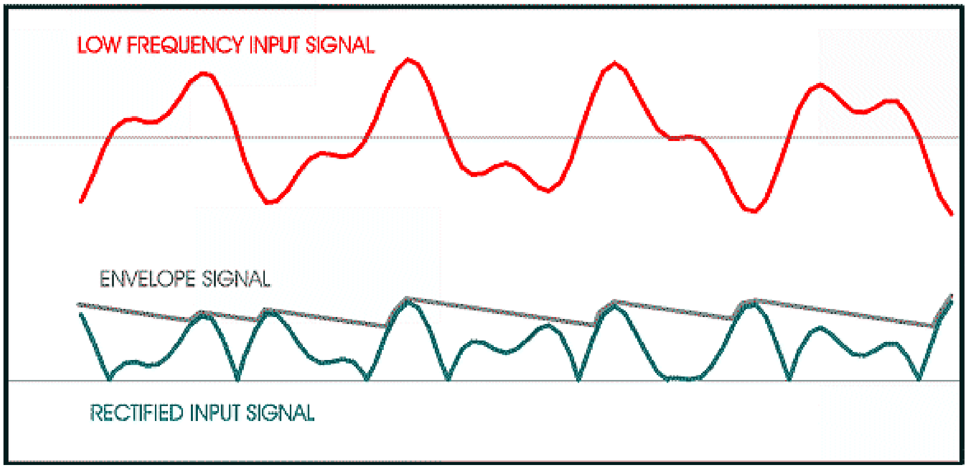


Figure 10

Figure 10 shows an example of the output of the envelope signal. The envelope signal can be measured at the 47uF capacitor inside the envelope calculator circuit. I simply took a variable signal at the input and looked at both the input signal and the output signal of the envelope calculator at a 2-channel oscilloscope. It immediately worked as I hoped for, and as it didn't need any debugging I never developed any test procedure.

THE RESULTS

I recommend setting the psychoacoustic bass enhancer just by ear and good taste. No specific instructions. Figures 5 and 8 show the effects of the Filter Threshold and Harmonic Spectrum controls respectively. People should experiment with the bass enhancer. If the Analoguer and the psychoacoustic bass enhancer are built together, remember that they are independent of each other. One works at the low frequency domain, the other at the high frequency domain. The Analoguer does have its own bass enhancement section, and it can be used with the psychoacoustic bass enhancer for best effect.

The following WAV sound clip demonstrates the effect of the psychoacoustic bass enhancer. The file is mono, 8-bit resolution to keep its size limited. There is some high-frequency noise which comes from my laptop PC. I did clear up some of the noise, but wasn't able to remove it completely. The file contains six test tones:

5 seconds 100 Hz sine wave
5 seconds 50 Hz sine wave
5 seconds 25 Hz sine wave
5 seconds 100 Hz sine wave + bass enhancer
5 seconds 50 Hz sine wave + bass enhancer
5 seconds 25 Hz sine wave + bass enhancer

Download bass enhancer demo WAV file (73Kbytes)

The first three sections will show how the 100 Hz can be easily reproduced by most loudspeakers/headphones whereas the 50 Hz already is less strong. The 25 Hz normally is inaudible. After passing the tones through the bass enhancer with the Filter Threshold set at position 1, the 100 Hz signal has hardly changed, the 50 Hz signal has a slightly changed timbre, and the 25 Hz becomes audible!

Note 1: I used the rather cheap microphone input of my laptop to create the files. Despite filtering some "noise" is still present.

Note 2: To decrease file size the wav-file has a sampling rate of 22 kHz and a resolution of 8-bit.

My Sennheiser HD600 headphones are said to have a deep and tight bass. However, testing with a frequency generator revealed that hardly anything happens below 30~40 Hz. These phones are simply not able to reproduce these low frequencies, even with the volume cranked up. However, when I turned up the potentiometer of the bass-enhancer things changed dramatically. I even could feel and "hear" tones down to 20 Hz wobbling my eardrums! Not surprisingly these tones tend to have a relatively "light" color, but they were definitely recognized as 20 Herz tones and not 40 or 60 Hz. Frequencies between 30 and 50 Hz were reproduced remarkably well, sounding round and "weighty".

The amount of feedback in the harmonics generator did have a notable influence on the sound. Personally I preferred the 22k ohm settings. If you do not like to add a switch with various feedback values I suggest using this value

Testing with real music first revealed that only little music has substantial frequency components below 50 Hz. Sure, certain organs go down to 16 Hz. and a piano grand goes down to 27.5 Hz but these tones are very rarely used. Actually, I'm a piano-player myself and I only know of one piece that uses the lowest octave; a piece by Bartok called "With Drums and Pipes" (fantastic music by the way, from the piano suite "Out of Doors").

So, with most music, my impressions were not staggering at first hearing. There was only a rather subtle effect, if any at all. However, after selecting pieces that really go deep, the effect was found to be most satisfying. Suddenly I became aware of an acoustic environment in which the deeper tones developed, a texture from which the lowest frequencies evolved in a most natural way. As a result one simply gets drawn more into the musical scenery.

Of course I also tested with loudspeakers. A friend of mine has Quad ESL63 loudspeakers. Very nice indeed, but due to their working principle, little sound pressure is present below 60 Hz. Imagine how he looked when I suddenly made a 25 Hz note audible! Again tones below 60 Hz sounded very substantial, albeit a little bit light in color. Testing with music in speakers confirmed my experiences with my headphones. The effect normally is very small but if it’s there, it really can be very involving.

People might argue that the bass tones produced are artificial and never can represent real uncolored bass tones as produced by big and mighty loudspeakers. However, there are many arguments for the psychoacoustic bass enhancer. First of all, big loudspeakers do not produce a real uncolored bass. A 30 Hz tone has a fundamental wavelength of 11 meters. No such frequencies can be properly reproduced in a normal-sized living room due to cancellation effects. Reflections to walls, floor and ceiling will cancel a large part of the original sound waves. We do feel the deep bass tones since everything starts to shake, but hearing is a different thing.

Moreover, room resonances result in a very uneven frequency response and the decay time of the acoustic energy is very long. The major problems with room acoustics generally are found in the lowest bass region, not in the treble (unless you listen to music in your bathroom). The psychoacoustic bass enhancer shifts the acoustic energy into a frequency region where room acoustics are much less problematic. The music is heard and not merely felt.

In headphones, the distance between ear and driver is much smaller than the distance between ears and walls and therefore the reflected waves are much smaller. Of course there are reflections at the skin, etc., but due to the long wavelength, reflected waves and direct wave are in phase. Only at very high frequencies does this cavity influence sound characteristics by interferences.

I’m aware that my description of the circuitry is rather short and that at a first glance you will come up with many questions. That’s on purpose . If you’re not able to answer most of these questions by yourself you will not be able to build this device.

Have fun,

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