ELECTRONOTES

WEBNOTE 44 10/24/2016

ENWN-44

ON DEMODULATION AND SUPERPOSITION

There has been interest here [1] in “hypersound” as a beamed source of modulated untrasound that carries and distributes ordinary audio. The basic concepts are not new: Beaming of sound (audible or ultra) is not new. Modulation is not new, nor is demodulation. [In fact, demodulation by a non-linear process is not new – that’s how even the original “crystal-set” radios worked.] What is new is a practical realization that seems to work (but initially seemed unlikely), and possible suggestions that it could be extended to beams of microwaves.

Fig. 1 shows a modulation example – ordinary Amplitude Modulation (AM) as was first used in radio. Here the top panel is the carrier, a frequency of 1/10. It is a sinusoid but is impossible to see in the clutter of so many cycles. It corresponds to the Radio Frequency (RF) carrier, like 550 kHz to 1,600 kHz for AM radio. The middle panel is the “program” or modulating signal, which would be Audio Frequency (AF) in AM radio. Note well that while it is AF, it is NOT audio (sound), but rather an electrical signal (usually a voltage) that is an analog of the sound (typically obtained by a microphone). Here the program signal varies from 0 to +1, so when we multiply the top two panels (giving the bottom panel), the modulation is 100%.

Here is the Matlab code for this plot:


```
t=0:4999;
xm=(1+sin(2*pi*t/1000))/2;
```

```
xc=sin(2*pi*t/10);
x=xc.*xm;
figure(1)
subplot(311)
plot(t,xc)           %Top panel, Fig. 1
subplot(312)
plot(t,xm)           %Middle panel, Fig. 1
subplot(313)
plot(t,x)            %Bottom panel, Fig. 1
figure(1)
```

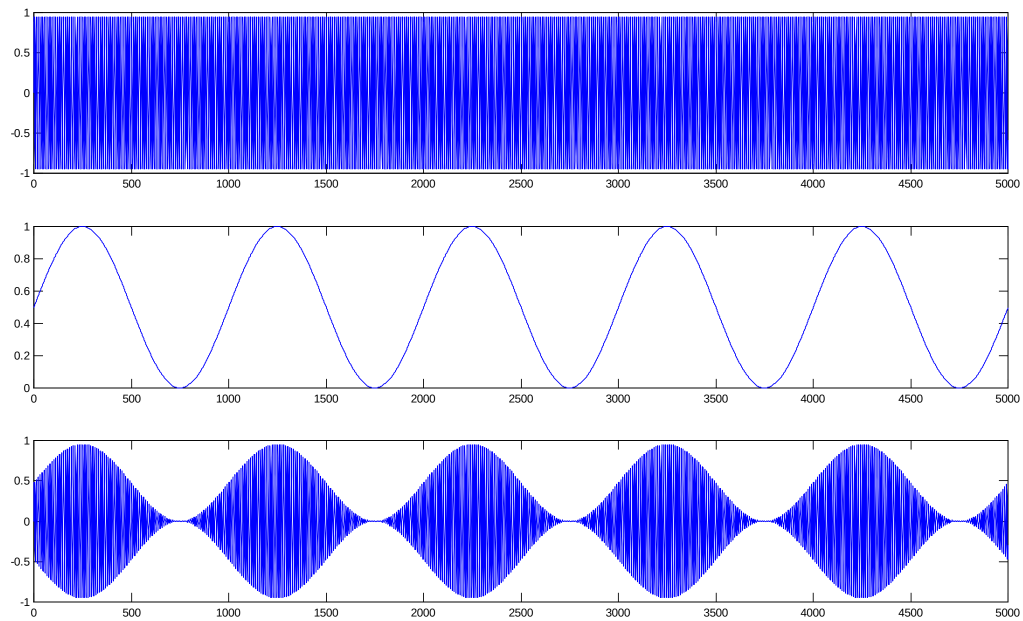

Fig. 1 Ordinary AM

In AM radio, it is the modulated carrier that is intercepted, and we desire to recover the electrical signal (de-modulate) and convert that back to sound with an earphone or loudspeaker. The top panel of Fig. 2 is just the absolute value of the bottom panel of Fig. 1 (thus the rectified modulated carrier) while the lower panel in Fig. 2 is a low-passed version of the top. The filter is just a simple length-10 moving average for illustration. This does a good job of recovering the program signal. In the crystal-set radio, the rectifier was a galena crystal (a diode) and the low-pass was the mechanical inertia of the transducer (the diaphragm of the earphone) and/or the frequency limits of the ear (the earphone performing a dual function of converting electrical to audio and of low-pass filtering).

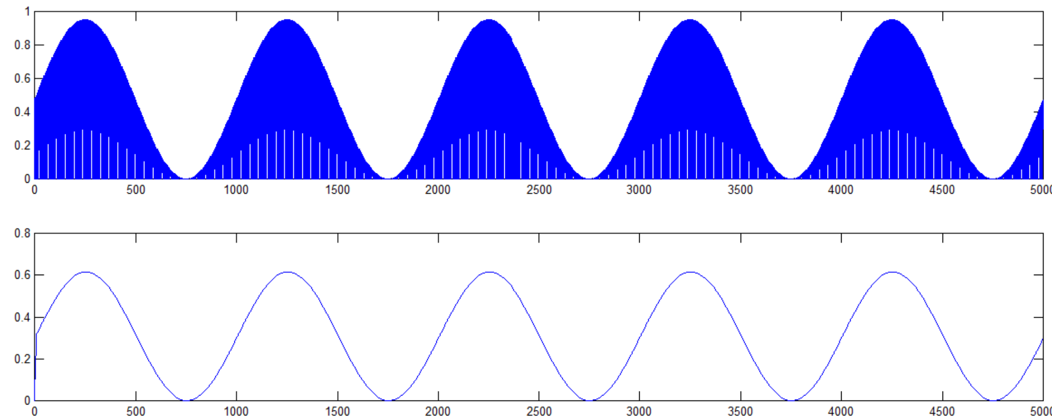

Fig. 2 De-Modulation with Rectifier/Low-Pass

The Matlab code for Fig. 2 is here:

```
%
xd=abs(x);
xdlp=filter((1/10)*ones(1,10), 1, xd);      % length 10 moving average
figure(2)
subplot(311)
plot(t,xm)
subplot(312)
plot(t,xd )                 % Top panel of Fig. 2
subplot(313)
plot(t,xdlp)                % Bottom panel of Fig. 2
figure(2)
```

Finally we want to illustrate the de-modulation possible with a non-linearity. [Wait a minute – we just did. The absolute value is a sever non-linearity.] It is common practice to investigate a nonlinearity startling with a power series. Here we will use x replaced by $x + 0.1x^2$. This is shown in the top panel of Fig. 3. Note the slight upward displacement of this relative to the modulated waveform (bottom Fig. 1) but far less than the absolute value (top Fig. 2). It is probably clear enough that if we only used x^2, we would have a purely positive result. The low-pass filtered (same filter as above) version of the non-linearity is shown in the bottom panel of Fig. 3. [Here, and above, the low-pass filter amplitude depends on several factors not central here.] The bottom panel is noisy-looking, but does suggest that credible de-modulation has occurred.

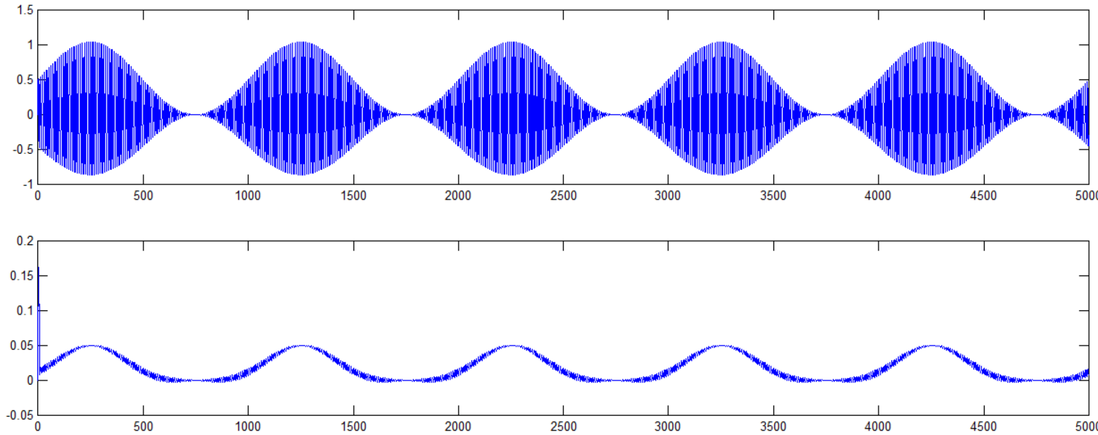

Fig. 3 De-Modulation with Non-Lineariaty (x → x + 0.1x^2)

```
The code for Fig. 3 is here:
%
xnl = x + 0.1*x.^2;
xnllp=filter((1/10)*ones(1,10), 1, xnl);
figure(3)
subplot(311)
plot(t,xm)
subplot(312)
plot(t,xnl )                    % Top panel of Fig. 3
subplot(313)
plot(t,xnllp)                   % Bottom panel of Fig. 3
figure(3)
```

If we interpret the results of Fig. 3 as being an ultra-sonic carrier and an audio program signal, we can enquire about the nature of the processes. The modulated (AM) signal would be obtained by using the audio signal (electrical analog from a microphone, etc,) to control the amplitude of an ultra-sound oscillator/transducer which then enters the air as ultrasound. We would expect this to be inaudible. We need a demodulator – in this case, the non-linearity. That is, the higher pressure of the assumed very loud ultra-sound produces a non-linear response. The corresponding low-pass filter is nothing more than the fact that the ear hears only the lower component caused by the non-linearity. The air is its own transducer (back to sound) here.

The "beaming" (beam-forming) is all in the transmitting transducer array for the ultra-sound. The demodulation is due to the high loudness of the ultrasound (non-linearity – requiring a lot of energy). No "transducer" is required, since the lower audio is there for the ear to interpret directly. High marks for those who actually made this work. Note as well that the difficulties were in engineering – not particularly with theory.

What About Microwaves?

Can an audio signal modulate a microwave beam? Of course: for decades (before fiber) this was how long-distance telephone was implemented once the limitations of land-lines needed to be overcome. Can microwaves be amplified into a non-linear region. Certainly: the circuitry can be driven to "clipping". BUT- you can't drive the MEDIUM (the metaphorical "aether") to non-linearity the way you can air. And the medium is not going to be its own transducer. You need demodulation circuitry and electrical signals to sound. And it needs to be dedicated technology on the receiving end, and put there intentionally.

Could microwaves be transmitted, even beamed by a satellite in space? Of course. Likely much of your communications comes from this. It's low-energy stuff however. Can vast amounts of energy (intense beams) be beamed to thousands of individuals on the surface? Of courses not. How would you get enough energy (in space) to drive your disruptor? It's hard enough to get the TV pictures down even with very sensitive receivers and lots of signal processing on the ground.

Hypersound does not translate to microwaves.

SUPERPOSITION

(added 10/24/2016)

(Of Signals – YES, Of Perceptions – Probabaly Not)

In the example above, we examined a case involving 100% AM. We are familiar with the notion of representing modulated signals in terms of discrete frequencies called "sidebands". This relates, for example, to the multiplication of two sinewaves as the absolutely equivalent sum of two different sinewaves [equation (1) below]. They are two views of the same signal. The crucial thing is that, in this traditional view, the "spectrum" is found "by observation" as the summation. [The best known example of a discrete spectrum is the Fourier Series sum.]

When we bring up superposition, and specifically inquire about what is "heard" when we superimpose waveforms AND PRESENT THEM TO THE EAR, we need to take great care. The perceptions are not necessarily (or perhaps ever) exactly superimposed. One example will serve at this point. Suppose we play a sinewave of 600 Hz to the ear. The spectrum is one frequency (600 Hz), and we hear a pitch at 600 Hz. If we then play a sinewave of 400 Hz, the spectrum is one frequency (400 Hz) and we hear a pitch of 400 Hz. If we add the two signals, the spectrum now has two frequencies (400 Hz and 600 Hz) but the ear <u>hears a pitch of 200 Hz</u>. This is the "missing fundamental" phenomenon. It is lower than either the two components. It leave a very different impression. Now, when you try hard (perhaps starting with an unpaired tone) it is quite possible to "hear out" the 400 Hz and 600 Hz components. The superposition is the sum of the parts (mathematically, electronically) but quite a bit more than the sum of the parts at the ear. <u>We need to listen and not guess</u>.

Below is Matlab code for playing this example:

```
n=0:14999;
fs=10000
x1=sin(2*pi*400*n/fs);
x2=sin(2*pi*600*n/fs );
superpos=x1+x2;
figure(6)
plot(n(1:500)/fs,s(1:500))
sound(x1,fs)
pause
sound(x2,fs)
pause
sound(superpos,fs)
```

At this point, three simple equations (just high-school trig) can be studied. Equations (1) and (2) are absolutely the same (balanced modulation) if we make the substitutions of variables. [The superposition is of two frequencies.] Equation (3) is just a modification of equation (2) for the case of 100% AM. This third case has three frequencies: the carrier (X) and two sidebands of half that amplitude, at (X+Y) and at (X-Y), Y being the program frequency. Whether we form (synthesize) these signals electronically by multiplication, or by summation, makes no difference to the final result.

$$sin(A) + sin(B) = 2\,sin\left(\frac{A+B}{2}\right)\cos\left(\frac{A-B}{2}\right) \qquad (1)$$

SUPERPOSITION

$$sin(X)\cdot cos(Y) = (1/2)\,[\,sin(X+Y) + sin(X-Y)] \qquad (2)$$

BALANCED MODULATION (Multiply, of Double Sideband)

$$sin(X)\cdot[1 + cos(Y)] = \sin(X) + (1/2)\,[\,sin(X+Y) + sin(X-Y)] \qquad (3)$$

AMPLUTUDE MODULATION (100%)

When we state that the signal is to be converted to sound, we eventually need to involve the nature of air as a propagating medium, and the way the ear perceives various pressure waves? For low to moderate levels of sound (relative to the ear), and for all frequencies in the audible range, the indifference of the result to mathematical interpretation and synthesis, with regard to summation of multiplication, remains.

Thus first here we consider a modulated signal in terms of a summation synthesis, where the demodulator is either absent, or is a non-linearlty, and that a simple low-pass filtering stands in for the transducer and/or for the ear.

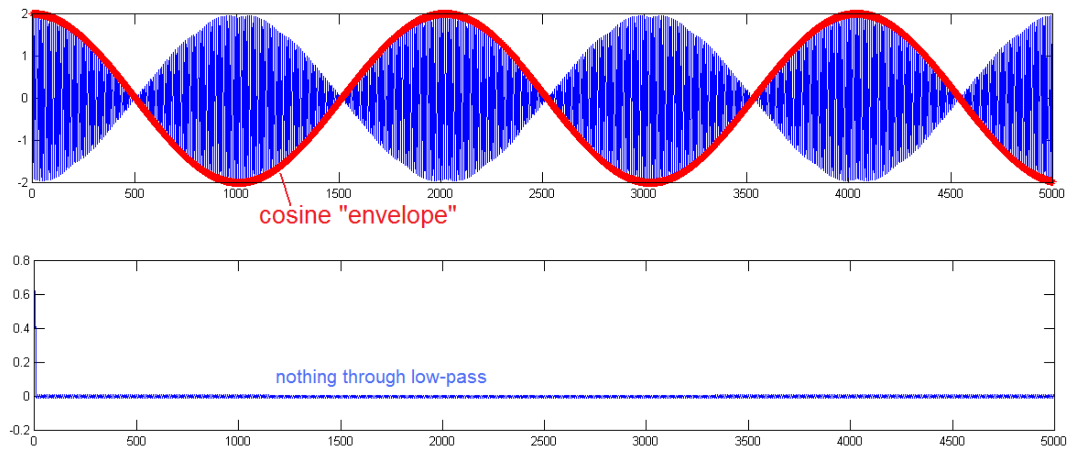

Fig. 4 Sum of Two Sinwaves - Beating

Here is Matlab code for Fig. 4

```
t=0:4999;
xm=sin(2*pi*t/10.1);
xc=sin(2*pi*t/10);
xe=2*cos(2*pi*(1/10-1/10.1)/2*t);          % calculated envelope
xtrig=sin(2*pi*(1/10+1/10.1)/2*t).*xe;
x=xc+xm;
xnl = x;
xnllp=filter((1/10)*ones(1,10), 1, xnl);
figure(4)
subplot(311)
plot(t,x)
hold on
plot(t,xe,'r*')
hold off
subplot(312)
plot(t,xnl )
subplot(313)
plot(t,xnllp)
figure(4)
```

Fig. 4, top panel, shows the sum of two sine waves which are added (xc+xm) and we see the lobed structure of balanced modulation. Also plotted, calculated as xe, in heavy red, is the cosine term which is the envelope of the beating waveform [2]. The average frequency, half the sum, is (1/10+1/10.1)/2 = 0.099595 and half the difference frequency is (1/10-1/10/1)/2 = 0.00495. The signal is thus a high frequency nearly equal to the components subjected to a dominating low frequency amplitude beat. Very different from either component.

The lower panel of Fig. 4 shows the low-passed version of the top panel. That is, the balanced modulation has no DC component (as is evident from the trig identities). Thus while there is a spectral component at (near to – actually) the higher frequencies (the average at about 1/10), there is no spectral energy at half the difference frequency (at about 0.005). There is a very strong perception that something is there, perhaps low audio, even sub-audio (as is the case of tuning musical instruments by beats). Thus while we perceive a low-frequency event, it is not found in the spectrum.

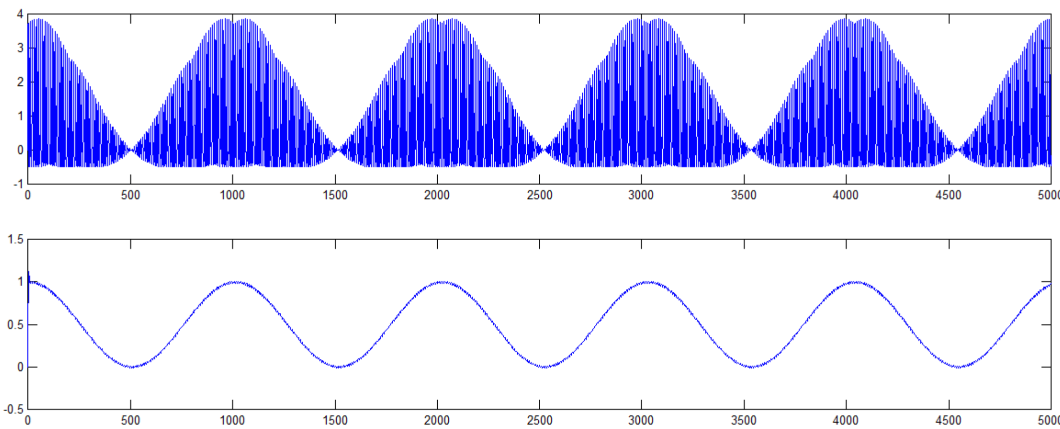

Fig. 5 Non-Linearity/Low-Passed - Demodulation

In Fig. 5 we again subject the signal to a non-linear operation. This is the same sort of demodulation we had in Fig. 3. Again we see that the symmetry of the waveform is disturbed (top panel), and some notion of a demodulated low-frequency sinewave emerges (lower panel). We may hesitate to call this a demodulation since we did not actually modulate anything, but just superimposed the two frequencies that would have been the double sidebands.

Here is Matlab code for Fig. 5

```
xm=sin(2*pi*t/10.1);
xc=sin(2*pi*t/10);
xe=2*cos(2*pi*(1/10-1/10.1)*t/2);
xtrig=sin(2*pi*(1/10+1/10.1)*t/2).*xe;
x=xc+xm;
xnl = x + 0.5*x.^2;
xnllp=filter((1/10)*ones(1,10), 1, xnl);
figure(5)
subplot(311)
plot(t,x)
hold on
plot(t,xe,'r')
%plot(t,xtrig,'c:')
hold off
subplot(312)
plot(t,xnl )
subplot(313)
plot(t,xnllp)
figure(5)
```

DISCUSION:

Here we have involved two topics: demodulation and superposition. Also involved is the common idea that non-linearities needs to be considered. In engineering, the ability to superimpose is closely associated with LINEARITY. With perception brought in, a valid claim of superposition, even without a non-linearity, needs to be demonstrate as it may well be absent.

REFERENCES

[1] the "here" is a reference to the website

http://thehum.info/

which had comments on various topics (October 2016), including possible beaming of ultrasound and/or microwaves purportedly as a causative element of "the Hum". The first part of this note involving demodulation by a non-linearity was a comment posted there on Oct. 17, 2016. Subsequently a discussion of superposition issues was added and became this Webnote.

[2] B. Hutchins, "Revisiting: Beating", Electronotes, Volume 22, Number 213 January 2013 http://electronotes.netfirms.com/EN213.pdf