SETTING THE TONE

In the last instalment of Workshop we looked at the creation of digitally synthesised sound using a digital-to-analogue converter, and designed a machine code program to generate three types of waveform: square, saw-tooth and sine waves. Now we look at two other important sound parameters: volume and pitch.

The volume of a tone is determined by the range of oscillation of the waveform generating the tone. In other words, volume depends on the difference between the maximum value of the waveform and the minimum value. This property of a sound wave is called the *amplitude.*

Using a simple BASIC program to oscillate values placed in the user port register we can demonstrate how easily amplitude can be controlled in a digital waveform.

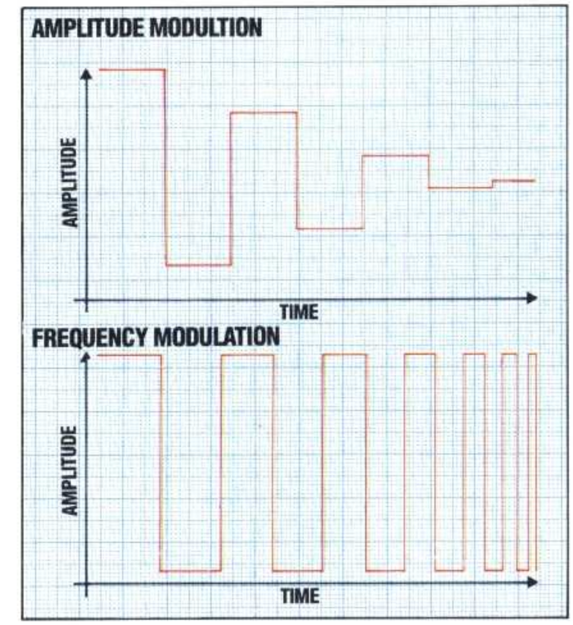
```
10 REM **** CBM BAS FRED/AMPLITUDE ****
20 :
25 DDR=56579:DATREG=56577
30 POKE DDR,255:REM ALL OUTPUT
40 FOR I=255 TO 0 STEP-15
50 FOR J=1 TO 100
60 POKE DATREG,I:POKE DATREG,0
70 NEXT J,I

800 REM **** SAMPLE PROGRAM ****
805 :
806 UP$=CHR$(145)
810 DIV=49798: REM AMPLITUDE FACTOR LOCATION
820 DEL=49799: REM DELAY FACTOR LOCATION
830 TME=49800: REM DURATION FACTOR LOCATION
840 CALL=49801:REM PROGRAM START ADDRESS
850 :
860 DDR=56577:POKEDDR,255:REM ALL OUTPUT
870 :
880 PRINTCHR$(147):REM CLEAR SCREEN
890 PRINT:INPUT"AMPLITUDE FACTOR 0-7";AF
900 IF AF<0 OR AF>7 THEN PRINT UP$;UP$::GOTO890
910 POKE DIV,AF
920 :
930 PRINT:INPUT"DELAY FACTOR 1-101";DF
940 IF DF<1 OR DF>101 THEN PRINT UP$;UP$::GOTO930
945 POKE DEL,DF
950 :
960 PRINT:INPUT"DURATION FACTOR 0-15";TF
970 IF TF<0 OR TF>15 THEN PRINTUP$;UP$::GOTO960
980 POKE TME,TF
990 :
1000 SYS CALL
1010 GETA$:IFA$="" THEN 1010
1020 IFA$="X" THEN 880:REM RESTART
1030 GOTO 1000REM ANOTHER BEEP
```

At the start of the program a crude square wave is generated that oscillates between 255 and 0. This means that the amplitude of the wave is 255. As the program runs, the upper value placed in the data register is reduced in steps of 15. As the upper value is decreased, so the amplitude decreases — and the effect of this, when monitoring the sound produced through a stereo amplifier or headphones, is that the volume of the tone gradually fades away to nothing. So the volume of a digitally synthesised tone can be controlled by limiting the range of values placed in the user port data register.

The pitch of a note is governed by the frequency of the generating wave; that is the number of waveform cycles per second — the larger the number of waveforms produced per unit time, the higher the pitch of the note heard.

Modular Construction
Waveforms can be modulated either by frequency or by amplitude. Frequency alters the pitch of the tone heard and is determined by the number of waveform cycles output per second. Amplitude alters the volume of the tone and is the difference between the maximum and minimum values in a cycle. The diagrams show how amplitude can be modulated so that the tone produced gradually decreases in volume and how frequency can be modulated to produce a tone which rises in pitch



Frequency can be controlled digitally in two main ways. The first is to increase frequency from a bottom limit by taking fewer samples of the waveform. If a wave were split into 100 samples, for example, then a machine code program would take a certain length of time to place each value in succession into the data register, and so a certain number of complete waveforms could be produced per second. The number of samples obviously governs the frequency of the tone heard. To double the frequency, the machine code program could, instead, take every second value only from the data table defining the wave. The frequency could be tripled by taking every third value, and so on. There are two drawbacks to this method. The first is that small frequency adjustments are difficult to make without distorting the shape of the wave. Secondly, as the frequency of the wave increases, the wave generated bears less and less relation to the original wave shape as fewer samples are used.

An alternative method is start with a loop that steps through the waveform data as fast as possible, thus providing a maximum frequency. Frequency can then be adjusted by inserting short delays in the loop. This gives us much more accurate control over the tone frequency, but means that the number of samples making up the waveform must be small if a reasonably high maximum frequency is to be obtained. We shall employ the second of these two methods to