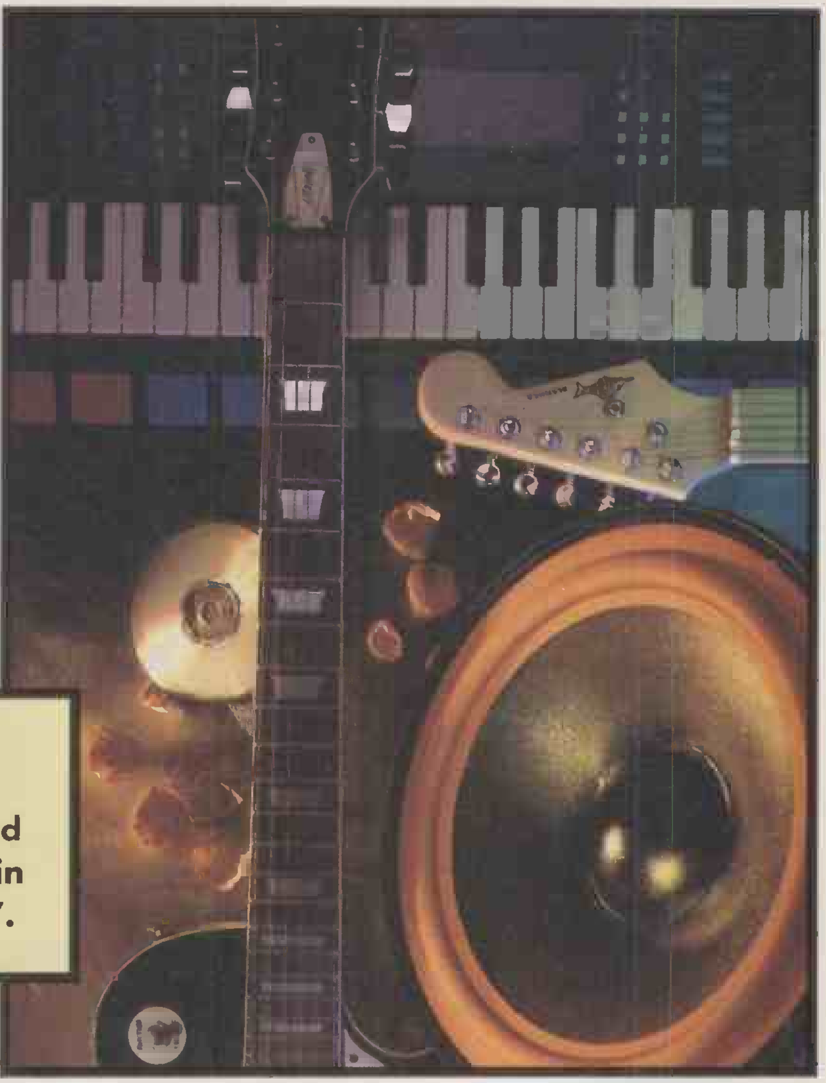

Richard Brice looks at various ways in which intentional electronic signal distortion is used to enhance music – article three in the series 'Music and electronics'.

Electronic effects in music

The contribution of electronics to the recording and reproduction of music is not solely limited to its capacity to transmit the musical performance. Electronics engineers have contributed – and continue to contribute – the means by which modern music is created. This is an act which is a part of the common ongoing creative enterprise we call 'art'.

I don't think it's stretching the truth too much to say that a host of today's trade names will become the Stradivarius or Broadwood of tomorrow. Nowhere is this contribution clearer than in the design of electronic musical effects.

Echo and reverberation

A true echo is only heard when a reflected sound arrives a twentieth of a second or more after the direct sound first reaches our ears. Compare that with the sound that accompanies the voice of a priest or of a choir as their effusions stir the roar of reverberation in the atmosphere of a vast, medieval cathedral.

Reverberation is made up of echoes too, but by a mass of echoes following more swiftly than those of a discrete echo.

Clearly most recording studios are not large enough for an echo to be a natural consequence of their design. Neither are most cavernous enough to possess the acoustics of a cathedral. And a good thing too for it is far easier to add artificial reverberation and echo electronically than it is to eliminate the natural form.

Electronics thereby underpins the philosophy embraced by most modern recording studio designers – aim for a dry natural acoustic and augment this with artificial reverberation, as required.

Artificial echo was originally accomplished by means of a tape delay device as illustrated in **Fig. 1**, the signal being fed to the record head and the 'echo' signal picked off the replay head which was situated separately and 'downstream' of the record head. The distance between the two heads and the tape speed determined the delay.

On commercial units, the tape speed was usually made continuously variable so as to realise different delay times. This arrangement, obviously, only produced a single echo. In order to overcome this limitation, to this simple device a circuit was added which allowed a proportion of the output of the replay head to be fed back and re-recorded. By this means was an infinitely decaying echo effect performed.

By altering the degree of feedback – known in this context as re-circulation – differing reverberant 'trails' could be achieved. Just as the early microphones had, in their time, fathered the vocal style of crooning – because they per-

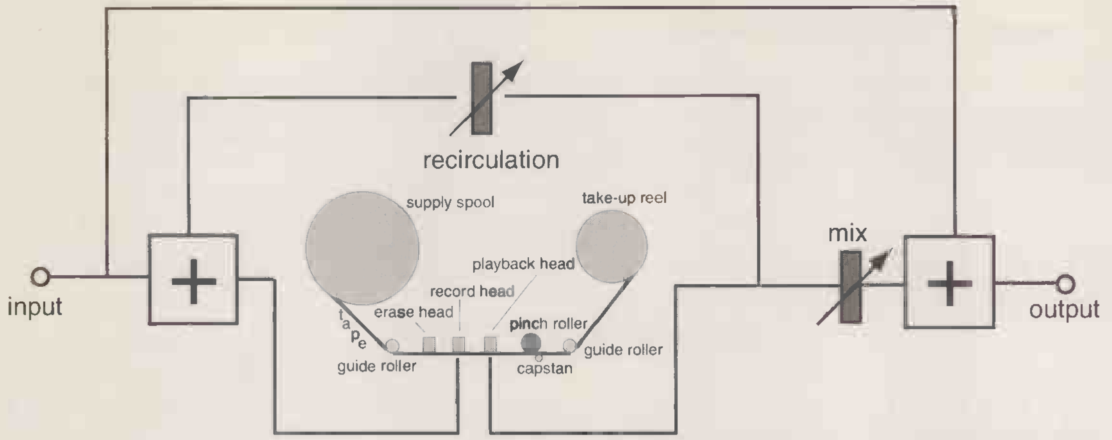

Fig. 1. Early machines produced an echo whose delay time was determined by tape speed and the gap between the record and replay heads. Next generation echo machines fed back some of the replay head output, allowing multiple echoes.

formed best when capturing slight sounds very close to the diaphragm – so the tape-based echo unit spawned an entire vocal technique too.

Modern digital delay devices have shunned tape techniques but accomplish the same effect by delaying suitable digitised audio signals written into, and read out of, a random-access memory store, **Fig. 2**. Alternatively hybrid digital/analogue techniques are utilised which exploit 'bucket-brigade' delay lines.

Both these techniques have all the obvious advantages of a purely electronic system over its electromechanical precursor. But there is one exception. Often, the rather poor quality of the tape transport system in the early devices introduced a degree of randomness – in the form of wow and flutter – into the replay system which help ameliorate a 'mechanical' quality which the resulting echo otherwise has.

Digital devices exhibit this quality quite distinctly – particularly at short delay times when the tail takes on a characteristic 'ring'. This unwanted outcome manifests itself more clearly still when the initial delay shortens. When a simple delay and re-circulation technique is employed to synthesise a reverberant acoustic, it can take on a very unnatural quality indeed.

Better results are obtained when a number of unequally spaced delay points, or taps, are used and these separate signals fed back in differing proportions, i.e. weightings, for recirculation.

Top quality delay and artificial reverberation units go so far as to introduce quasi random elements into the choice of delay taps and weightings so as to break up any patterns which may introduce an unnatural timbre to the artificial acoustic. Fortunately digital techniques have come so far that reasonable results are obtainable at very low cost. Artificial delay and reverberation are almost always incorporated in the audio system via the audio console effect send and return.[1]

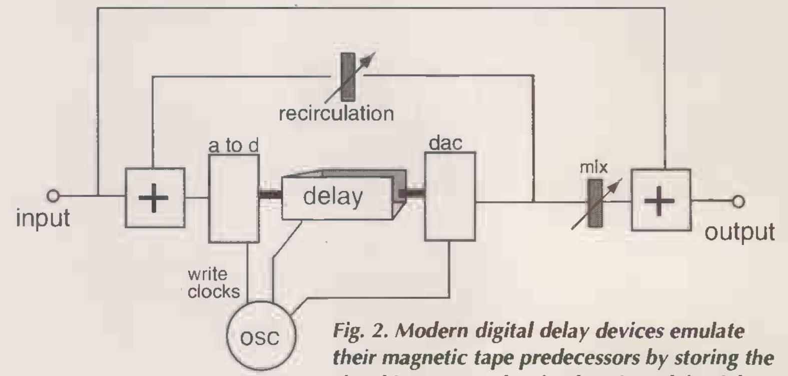

Fig. 2. Modern digital delay devices emulate their magnetic tape predecessors by storing the signal in memory for the duration of the delay time.

Table 1. Even harmonics tend to be musically related to the original input tone, whereas high-order, odd harmonics are musically unrelated to the original guitar signal.

Harmonic	Musical note	Comment
Fundamental	C	
2nd (1st overtone)	c	octave
3rd	g	twelfth (octave + fifth)
4th	c'	fifteenth (two octaves)
5th	e'	seventeenth (two octaves + major third)
6th	g'	nineteenth (two octaves + perfect fifth)
7th	b-flat' (nearest note)	dissonant; not in natural scale
8th	c"	three octaves
9th	d"	major 23rd (three octaves + second)
10th	e"	major 24th (three octaves + third)
11th	f" (nearest note)	dissonant; not in natural scale
12th	g"	major 26th (three octaves + fifth)
13th	a"	dissonant; not in natural scale
14th	b-flat"	dissonant; not in natural scale
15th	b"	major 28th
16th	c"'	four octaves
17th	C#"'	dissonant; not in natural scale
18th	d"'	major 30th
19th	d#"'	dissonant; not in natural scale
20th	e"'	major 31st

Guitar amplifiers – distortion and fuzz

Usually an effect to be guarded against in both design and operation of any audio circuit is the severe amplitude distortion known as clipping. But for guitarists, this effect is amongst their stock-in-trade. 'Grunge' has re-established this sound in recent years.

Known variously as fuzz, overdrive or plain distortion, the manner in which the circuit overloads becomes an integral part of the sound timbre. So much so, that for guitarists, a whole mythology surrounds this subject.

The first commercially available unit intended for the purpose of generating severe waveform distortion was the Gibson Maestro *Fuzztone* amplifier.† In developing the *Fuzztone*, Gibson crossed something of a Rubicon in audio electronics design. Before it, the role of the audio amplifier was not intended to be a feature of the sound; indeed many of the amplifiers were intended to be 'distortionless'. But guitarists pushed the equipment to its limits in search of expressive potential and thereby uncovered corners of the performance-envelope unforeseen by the equipment's designers.

Ironically, often it was precisely at these boundaries that the greatest potential for artistic utterance was found, thereby establishing the sonic signature of a particular perfor-

† It was this unit that was used on the Rolling Stones record *Satisfaction* in 1965.

mance limitation as a *de facto* standard for acolytes and imitators alike.

In turn, manufacturers have been forced to continue to build equipment which is deliberately designed to expose a design limitation or else find a way of simulating the effect with more modern equipment. Hence the inclusion of the apparently objective subject of amplifier design in this article on creative effects.

Relieved of a duty to be accurate, instrumental amplification is very difficult to analyse objectively. However a few observations may be made with some certainty: Firstly, most amplification is not usually designed with a deliberately modified frequency response. This is more usually a function of the designer's choice of loudspeaker and housing. Amplifiers – both low-level and power-level – are more usually engineered for their distortion characteristics.

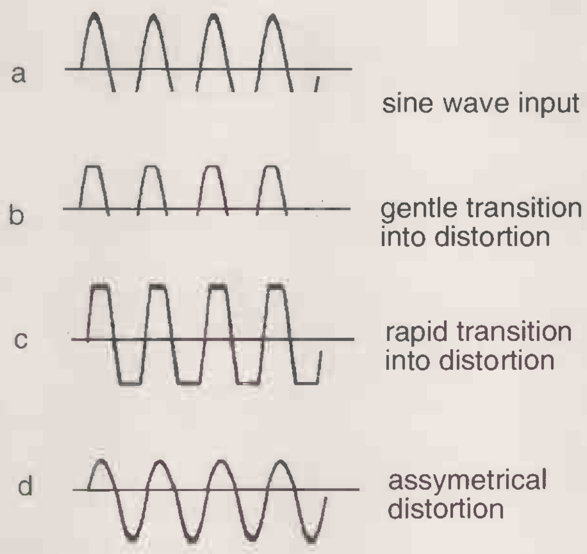

Fig. 3. From a) to d), original signal, sluggish transition into overload, fast transition into overload and finally, the result of an asymmetrically overloaded amp.

Difficult to see – easy to hear

Research has been done to connect various transfer curve characteristics with subjective perceptions. Once again the ear proves to be a remarkably discerning apparatus. So that, in spite of the fact that all distortion mechanisms perform roughly the same 'function', each commercially available amplifier has its own distinctive sound and loyal adherents – some units having acquired an almost cult status.

While many of these differences might be difficult, if not impossible, to analyse, a number of distinguishing characteristics are obvious enough. Firstly, the forward gain of the amplifier has an effect on the rate of discontinuity between the linear and non-linear portions of the transfer characteristic.

A unit with a low forward gain and a small degree – or no – negative feedback will show a sluggish transition into the overload region. Such a unit will produce a distortion on a sine wave input like that illustrated in **Fig. 3b**. On the other hand, unit with a high forward gain and a good deal of negative feedback, and thus a faster transition into the non-linear region, will produce an output waveform more like that shown in **Fig. 3c**.

Secondly, the character of the distorted sound depends to a large measure on the degree of asymmetry imparted to the output waveform by the overdriven amplifier stage. An asymmetrically distorted waveform – like the one in **Fig. 3d** – has a far higher proportion of even harmonics than the waveform shown in Fig.3c, which has a high proportion of odd harmonics.

Even harmonics tend to be musically related to the original input tone, whereas high-order, odd harmonics are musically unrelated to the original guitar signal, **Table 1**. This suggests that an amplifier producing a symmetrical overload characteristic will tend to sound 'harsher' than a unit yield-

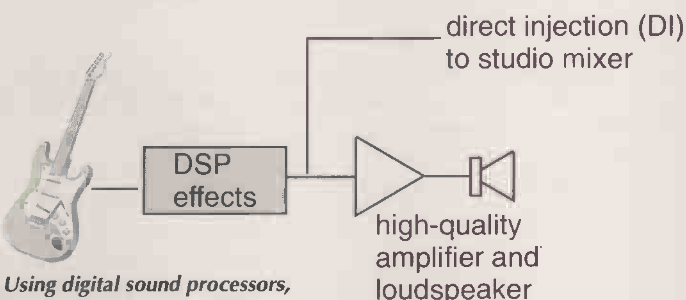

Fig. 4. Using digital sound processors, distortion can be carefully controlled by passing the linear pulse-code modulated signal through a look-up table stored in read-only memory with any desired transfer-function.

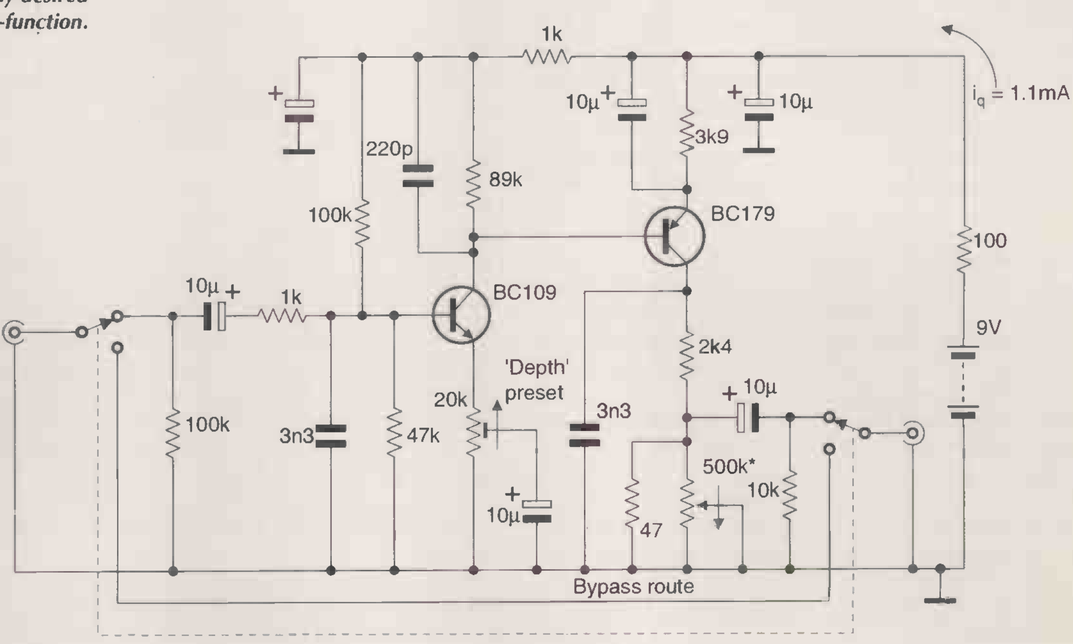

Fig. 5. There are complex digital circuits for producing distortion, but simple analogue alternatives like this one are often preferred.

ing asymmetrical distortion, and subjectively this is indeed the case.

Why valves?

Valve amplification is almost certainly preferred for its asymmetrical transfer characteristic and for its longer transition band from 'non-distorting' to 'distorting' regimes. This gives the instrumentalist a wider and more controllable tonal and expressive palette. This characteristic is enhanced by very limited amounts of negative feedback.

More elaborate semiconductor-amplifier counterparts, due to a high level of derived linearity and very high forward gain, tend to elicit a rasping, strident tone when in overload. Unfortunately, the designer has little or no option when faced with the design of a solid state amplifier.

Being of essentially Class-B design, these amplifiers cannot function without large amounts of negative-feedback therefore the designer of such an amplifier is forced to adopt up-stream electronics to try to emulate the gradual distortion characteristics of a valve amplifier.

With the advent of digital electronics, this philosophy has blossomed. Inside digital sound processors, distortion can be carefully controlled by passing the linear pulse-code modulated signal through a look-up table stored in read-only memory with any desired transfer-function – **Fig. 4**.

Nevertheless analogue alternatives are often preferred and may be extremely simple. A design which has been used for some years, and which has appeared on many professional recordings, is illustrated in **Fig. 5**. Effectively the transistor pair creates a high gain amplifier – enough to drive the output signal well beyond the supply rails.

The collector load on the second stage is split. This reduces the overall gain back to around unity and provides an adequately low output impedance. Control of the ac emitter load of the first transistor alters the gain of the amplifier and therefore the depth and character of the distorted output.

Wah-wah

Wah-wah is a dramatic effect derived from passing the signal from the electric guitar's pickup through a medium-Q band-pass filter, the frequency of which is adjustable usually by means of the position of a foot-pedal as illustrated in **Fig. 6**.

The player may the use a combination of standard guitar techniques together with associated pedal movements to produce a number of instrumental colours from an almost percussive strumming technique to a lead guitar style – usually in combination with fuzz effect – in which the guitar almost 'cries' in a human-like voice.

Pitch shifting

Pitch shifting is used for creating 'instant' harmony.

Simple pitch shifters create a constant musical interval above or below the input signal. You might think that such a limitation was pretty devastating. However, various automatic transpositions produce acceptable results. For instance, a harmony at a perfect fifth produces the scale in **Fig. 7**. This scale is usable except for the F-sharp.

Harmony at a perfect fourth is even better, **Fig. 8**. It has only one note that is not present in the key of C major, like the harmony at the perfect fifth. But the note is B flat which is a prominent 'blue' – i.e. blues scale – note in C major. For this reason, it is often acceptable in the context of rock music.

The instant transpositions of perfect fourth up – or its lower octave equivalent, perfect fifth down – are the most common transpositions employed in simple pitch shifters, with the exception of octave transpositions. Guitarists in particular most often employ a pitch shifter in one or other of these two roles.

Table 2. For pitch shifting, natural ratios of input versus output clock are preferred since they are related by simple numerical ratios.

Interval	Frequency ratio
Octave	2 : 1
Fifth	3 : 2
Fourth	4 : 3
Major third	5 : 4
Minor third	6 : 5
Major sixth	5 : 3
Minor sixth	8 : 5

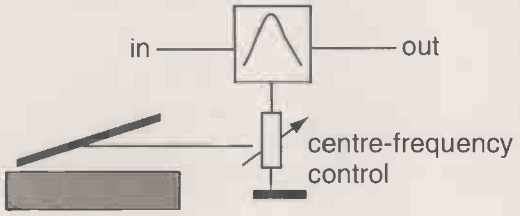

Fig. 6. Normally, a foot pedal is used to control the wah-wah effect's centre frequency.

Fig. 7. Scale representing harmony at a perfect fifth.

Fig. 8. Harmony at a perfect fourth is even better than that of the perfect fifth in Fig. 7.

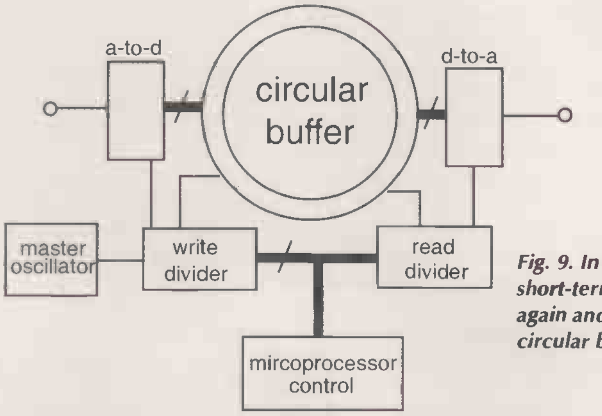

Fig. 9. In pitch shifting, the short-term store is re-used again and again, i.e. it is a circular buffer.

Intelligent pitch shifters can be programmed to produce a harmony related to a selectable musical key, so that a musical harmony can be created.

Technically, pitch shifting is achieved by converting the input signal to a pulse-code modulated digital signal, writing audio data into a short term store, and reading it back out at a different sample rate. Thereafter, the resulting pcm signal is converted back to an analogue signal. Because the short-term store is used over and over again it is referred to as a circular buffer, **Fig. 9**.

Pitch shifting by various musical intervals is achieved by adjusting the ratios of the input and output clocks.

Natural ratios are preferred for – as Pythagoras noticed two-and-a-half thousand years ago – these are related by

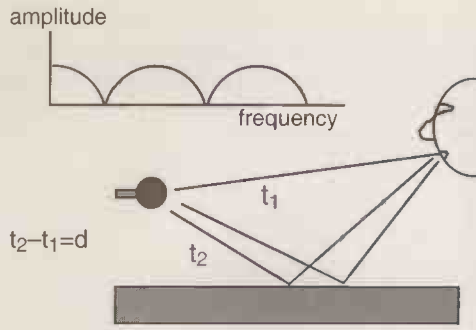

Fig. 10. A common problem encountered in microphone technique is the accidental establishment of multiple path lengths between sound-source and microphone element

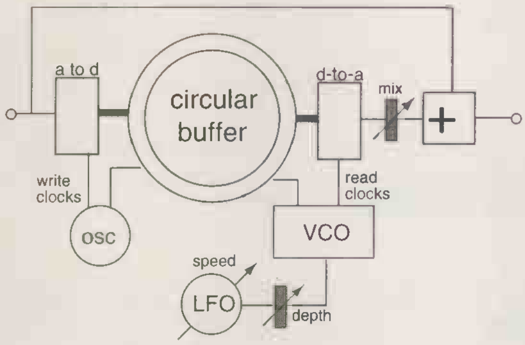

Fig. 11. Early flangers were based on magnetic tape, the flanging effect being produced by manually slowing one of two synchronous tape recorders. In this electronic version, a low-frequency oscillator simulates the hand.

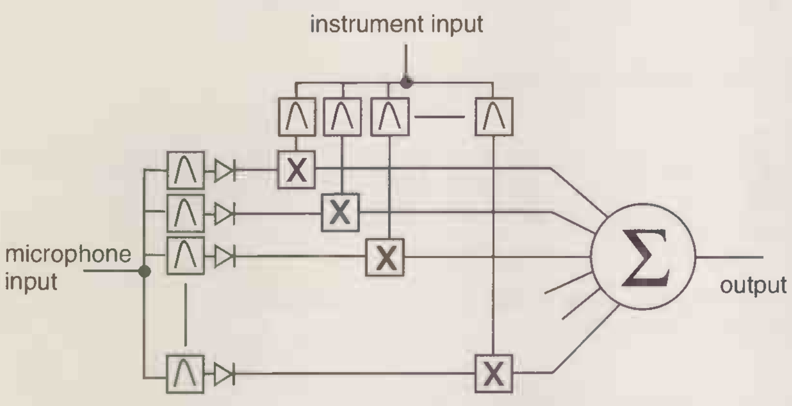

Fig. 12. The principal feature of the vocoder is its two inputs – one for an instrument and another for a microphone.

simple numerical ratios, **Table 2**. Note that most pitch shifters allow for a bypass route so that the original sound can be mixed with the harmony in a desired proportion before the signal leaves the unit.

Another application of the pitch shifting technique described in the last section is in creating audio effects which, to the uninitiated, sound nothing like pitch shifting as all. Instead effects such as chorus, flanging and so on, create a swirling, thickening texture to a sound.

Flanging, phasing and chorus

A common problem encountered in microphone technique is the accidental establishment of multiple path lengths between sound-source and microphone element, which creates a 'comb-filter' effect whereby successive bands of frequency are reinforced and cancelled, as illustrated in **Fig. 10**.

Although such an eventuality is extremely undesirable in the context of recording speech or sung sounds, the phenomenon produces an interesting acoustic effect – a kind of hollow ring.

Even more interesting is the effect as the microphone is moved in relation to sound source and reflecting body. This causes the frequency bands of reinforced and cancelled output to change. Imparting on the captured sound a strange, liquidity – a kind of 'swooshing, swirling' ring.

Of course, such an effect is not practically obtainable using moving microphones.* Instead it relies on utilising an electronic or electromechanical delay-medium to recreate an acoustic delay.

This effect has come to be known as flanging. It came about due to the slight lack of synchronisation between two 'locked' tape recorders; an effect which was exacerbated by rubbing a hand against the supply-reel flange of one of the tape recorders in order to slow it down in relation to the other.

A modern flanger dispenses with a tape mechanism to create the delay and, instead, a digital lag circuit is almost always employed. A low-frequency oscillator, or lfo, replaces the flange-rubbing hand.

The amplitude of the lfo signal controls the depth of the flange. This is equivalent to the amount of 'de-synchronisation', and it is controlled as shown in **Fig. 11**. The speed of the flange controls the frequency of the lfo and the degree of the effect is controlled in a mixing stage as shown.

Attempts at non tape-based analogue flange techniques involved the use of adjustable, cascaded all-pass filters providing the necessary delay elements. These circuits only produce a very small amount of delay per circuit. Even with a relatively large numbers of delays cascaded together, the delay was small in comparison to that required for a full flange effect. Because of this, these devices produced a particular, gentle effect, sonically apart and worthy of its own name – 'phasing'; a term based on the fact the circuits produce phase-shift, rather than full delay.

In a modern digital processor, the terms phasing and flanging really describe the same effect; the term phasing being used to describe very light degrees of flanging with delays up to about 1ms. Flanging uses delay variations in the region 1ms to 7ms.

Chorus is the next term in this continuum. In a chorus effect, the feedback fraction and the minimum delay-time are limited so as to ensure the depth of the comb-filter effect is much less pronounced than in the flanger or phaser. In a chorus effect the delay typically varies between 20 to 40ms.

Phasing and flanging and chorus find their metier in the hands of guitarists. Or should I say feet? Usually, this effect is incorporated in a pedal. This refinement facilitates switching the effect in and out without the guitarist having to take his or her hands off the instrument.

Vocoder

The vocoder is a device which allows the unique expression of the human voice to modulate an instrumental sound which may be monophonic or, more often, polyphonic. In order to understand the vocoder, it's worthwhile taking a few minutes to understand the production of vocal sounds.

The fundamental sound source involved in vocal production is a rather low frequency complex tone produced when air from the lung travels up the windpipe and excites the vocal folds in the larynx. The source of sound energy is known as the glottal source because the space between the vocal folds, and between which the air from the lungs is forced, is known as the glottis.

The vocal tract, comprising the pharynx (throat) the nose and nasal cavities and the mouth, subsequently modifies the spectrum of this glottal source. The vocal tract's shape can be varied extensively by moving the tongue, the lips and the jaw. In so doing, the spectrum of the glottal source is modified as it is filtered by the various resonances formed in the discrete parts of the vocal tract. Each of these resonances is known as a formant and each is numbered; the lowest frequency formant being termed the first formant, the next – the second and so on.

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* It is practical to generate such an effect by moving loudspeakers in relation to a fixed listening or microphone position. The *Leslie* loudspeaker works in just such a fashion.

The principal feature of the vocoder is its two inputs – one for an instrument and another for a microphone. The block diagram for a simple instrument is given in **Fig. 12**. Vocoder operation relies on the amplitude envelope of the vocal formants modulating the instrumental inputs via audio signal multipliers: these are voltage-controlled amplifiers in an analogue vocoder.

In circuitry terms this involves splitting the vocal signal and the instrumental signal into a number of frequency bands by means of band-pass filters. The greater the number of bands, the better the performance of the vocoder function. In a digital vocoder, the frequency spectrum can be split into a great many bands by means of a wave filter.

Following the band-dividing filters, the vocal signal path passes to a number of amplitude envelope-detector circuits, i.e. peak rectifiers in an analogue circuit). These envelope signals are then used as the variables applied to each of the multipliers following every band-dividing filter in the instrumental signal path. In this way, the frequency spectrum of the speech is 'imprinted' on the instrumental sound.

A physiological analogy

You can draw a physiological parallel by saying it is as if the lungs and vocal folds were replaced with the instrumental sound while the function of larynx, mouth and nasal cavities remain the same.

Not only is the Vocoder capable of some exotic 'colouristic' effects. An example of this is found in Laurie Anderson's *O Superman*. But the physiological analogy may also have suggested to you an application whereby the instrumental input can be a synthesised tone similar to that produced by the lungs and vocal folds.

If that synthesised tone – or tones – is under MIDI control, the vocoder can be used as an artificially enhanced voice - always in tune and able to sing in perfect harmony with itself. In this variation, the vocoder is worthy of the separate name – 'harmoniser'.

Note that the harmoniser is not a pitch shifter. If harmonisation was achieved as described above – and it can be for certain effects – the vocal formants would be transposed along with the pitch; producing an effect termed 'munchkinisation'.

Talk-box guitar effect

Lying somewhere between the wah-wah pedal and the vocoder is the talk-box guitar effect.

The talk box exploits the unique and expressive acoustic filter formed by the various resonances of the vocal tract and mouth to modify the sound of an electric guitar. This is done by driving a small loudspeaker with the amplified guitar signal, feeding it through a horn and into a plastic tube. The tube is then clipped or gaffer-taped up the microphone-stand into a position so that it can be fed into the mouth of the guitar player.

The resulting sound is recorded via the microphone feed. Talk boxes feature in the recordings of Aerosmith, Frampton and Joe Walsh among others. ■

Reference

1. Brice, R. (1988), Music Engineering, Newnes.