A Simple Ring-modulator

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I have lost count of the number of times that I have been asked to write down the basic circuit for a ring-modulator since building my first one early in 1968, when someone else suggested components for the circuit that I had found in two books. Up to now, I have constructed about a dozen ring-modulators, four of which were for my own use (two built in a single box for concerts, one permanently installed with my equipment at home, and one inside a footpedal with a controlling oscillator also included). So it seems to be helpful to make this information more available.

The ring modulator (hereafter referred to as RM) was originally developed for telephony applications, at least as early as the 1930s (its history has been very hard to trace), in which field it has subsequently been superseded by other devices. A slightly different form – switching type as opposed to multiplier type – occurs in industrial control applications, such as with servo motors. The RM is closely related to other electronic modulators, such as those for frequency and amplitude modulation (two methods of radio transmission, FM and AM, but now equally well known to musicians in voltage-controlled electronic music synthesizers). Other such devices include phase modulators/phase shifters and pulse-code modulators. All modulators require two inputs, which are generally known as the programme (a complex signal) and the carrier or control signal (a simple signal). In FM and AM broadcasting, for example, the radio programme is modulated (encoded) for transmission by means of a hypersonic carrier wave, and demodulated (decoded) in the receiver. Similar demodulation can be effected with RMs (see Klangumwandler), but does not appear to be possible with such fidelity to the original signal, nor is there much need for this in musical contexts. The commonest musical form of any audio-frequency modulation (AM, FM and RM) is vibrato, where a rapid fluctuation takes place between two adjacent levels of loudness, pitch or timbre; more pronounced forms are trills and tremolos (FM) and 'beats' between two pitches that are not quite in tune (AM).

The RM appears to have been introduced into electronic music around 1953. The earliest references that I have been able to find are to the electronic music studios at NWDR Cologne (1954, for a radio drama, and from 1956 in compositions), NHK Tokyo (1955) and RAI Milan (1956). The early development of all areas of electronic music (i.e. including musique concrète) from 1948 to mid-1950s was primarily concerned with tape treatments and superimpositions; it was only in the second half of the 1950s that filters and RMs became more common. By 1961, when a survey of the equipment in the world's major studios was compiled, about half of them possessed a RM.

RMs are incorporated in two other devices that have been used in electronic music. The Vocoder (Voice Coder, developed in the 1940s) was intended for telephony, to analyse speech and resynthesize it electronically (permitting research to be carried out in simplifying the sound content of a persons's voice and thus increase the capacity of existing equipment – without losing intelligibility and recognisability). The two constituent parts, for analysis and synthesis, are the Coder and the Voder (Voice Operation Demonstrator), and the RM occurs in the latter. In the late 1950s Vocoders were used in studios in Munich and Tokyo, and later in a live performance piece, "North American Time Capsule 1969" by Alvin Lucier. The second device, the Klangumwandler or Frequenzumsetzer (= Frequency shifter, also described as a single sideband modulator), was invented by Dr. Ludwig Heck at SWF Baden-Baden in 1955, and used in electronic music there and subsequently elsewhere. It consists of the following sequence: RM1, filter, RM2 (demodulator), low-pass filter (with two built-in oscillators). This gives a change of spectrum as well as of frequency, and is similar to the effect of a single RM but with less substantial timbral change and with **either** the sum **or** the difference tone – see below – suppressed. The same circuit, with discrete units, had been used experimentally in Cologne prior to 1955, and can be found in the score of Stockhausen's "Telemusik" of 1966 as the so-called 'Gagaku circuit' (named after the first music it transforms in the piece).

Like all other electronic modification devices, the RM functions in a way that is very similar to one aspect of acoustic sound, but considerably magnified. Just as filters relate to the small variations in timbre and colour that are obtainable with conventional instruments, the RM relates to combination tones (sum tones – rarely audible, first discovered by Helmholtz in the 1850s – and difference tones – first discovered in 1745, and exploited by Tartini nine years later as an aid to precise tuning of double stopping on the violin; subsequently employed by organ builders to produce the very lowest 32' and 64' pedal notes more economically with two shorter pipes). The output of a RM consists of the sum of the frequencies at its two inputs and the difference between them, rejecting the actual input frequencies. It is this rejection that distinguishes the RM from the AM (in which it is in other aspects a special form), since the latter retains the programme input. However, many of the results obtained with RMs sound similar to those from FM. The two components of the outputs (sum and difference tones) are known as side-bands.

Most applications of RM involve a complex sound such as that of one or more conventional musical instruments, modulated by a simple sound source such as a sine-wave. It is of course possible to modulate one complex sound by another, but this must be done with great care (especially for several instruments), otherwise the result tends to become very muddy and undefined. In certain instances even white noise, the most complex of all sounds, can be effective, in which case the other input will probably be discontinuous and articulate the output as a 'gate' (this is also possible with AM). Similarly, both forms of modulator can produce envelope shaping.

The operation of the RM can best be explained by a simple mathematical demonstration. In this we assume that a note with a normal overtone spectrum, such as that of most orchestral instruments, is modulated by a sine-wave (with no overtones). The frequencies of this overtone spectrum could of course also be be actual pitches in a major chord. If the sine-wave is 100Hz (= Hertz, cycles per second; around the bottom of the bass clef), and the fundamental (underlined) of the instrumental note is also 100Hz, we get:

programme +carrier	sum	difference		output
(etc.)	(etc.)	(etc.)		(etc.)
600Hz	700Hz	500Hz		700Hz
500	600	400		600
400	500	300	2×	500
300	400	200	2×	400
200	300	100	2×	300
$\underline{100}$ + $\underline{100}$	$\underline{200}$	$\underline{(0)}$	2×	$\underline{200}$
				100
				$\underline{(0)}$

The RM output looks very similar to the overtone spectrum of the original programme. However, since halving or doubling a frequency produces the octave below or above the original frequency respectively, it will be seen that he octave relationships between the overtones, in addition to their proportions of dynamic level (which become increasingly weaker the further they are from the fundamental), are already somewhat altered: the new fundamental (200Hz) is an octave higher – with below it, a weaker first overtone of the second inaudible fundamental 0Hz – but its first overtone (sum) is no longer an octave above it, but only a fifth; and so on.

By changing the frequency of the sine-wave, we come still further from the original timbre spectrum of the programme:

150Hz gives (no octave relationships!):-		(etc.)	
		750	Hz
		650	
		550	
	2×	450	
	2×	350	
	2×	$\underline{250}$	
		150	
	2×	$\underline{50}$	

400Hz gives:-		(etc.)	
		1000	Hz
		900	
		800	
		700	
		600	
		$\underline{500}$	
		$\underline{300}$	
	2×	200	
	2×	100	
		(0)	

If we now drastically alter the relationships of the fundamentals, with the sine-wave either at a considerably higher frequency (2000Hz) or at a far more dissonant interval from the instrumental sound (217Hz, instead of the unison, fifth and double octave of the previous examples), the results show greater changes:

2000Hz gives:-	(etc.)	
	2600	Hz
	2500	
	2400	
	2300	
	2200	
	$\underline{2100}$	
	$\underline{1900}$	
	1800	
	1700	
	1600	
	1500	
	1400	

217Hz gives:-	(etc.)	
	817	Hz
	717	
	617	
	517	
	417	
	383	
	$\underline{317}$	
	283	
	183	
	$\underline{117}$	
	83	
	17	

Neither of these retains any octave relationships, although if we considered one extra overtone in the programme from '(etc.)', the difference tone that resulted (1300Hz) would produce one. Thus with the high frequency carrier we see that the overtone spectra of the outputs become progressively squashed together, the higher the carrier is, and always in a distinctive, non-harmonic relationship – and this is what causes the sound of ring-modulation to become almost always instantly recognisable, and thus difficult to use effectively (just as, in electronic music, with tape loops, tape echo delays and now, with synthesizers, the slow downwards sweep of a low-pass filter treatment of a complex oscillator waveform). The higher the carrier, the more brittle the sound becomes. A mid-frequency carrier, when it is less consonant with the programme (as with 217Hz), gives a Dalek-like sound (similar distortions can also be obtained around 15-25 Hz). It is always possible to calculate the effect of RM on a mathematical basis, but the results are at times unpredictable.

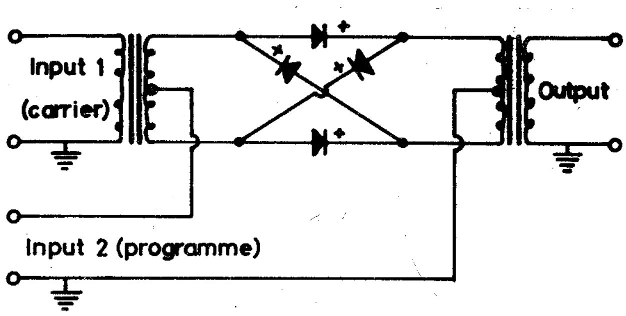
Well-known examples of the different possibilities of the RM in live performance are three works by Stockhausen: "Mixtur" (orchestra with four sine wave generators and RMs), "Mikrophonie II" (twelve singers, Hammond organ – as carrier – and four RMs), and "Mantra" (two pianos, two sine-wave generators and RMs). In "Mixtur" three areas of RM are exploited. Below c. 16Hz an oscillator is heard as pulses rather than as a continuous sound, and when such a frequency is the carrier, the programme is 'gated' in this **rhythmical** manner. With a medium-range programme, a 16Hz carrier will give combination tones that are approximately between a whole tone and one-eighth of a tone simultaneously above and below the programme pitches; thus between c. 10 and 20Hz the carrier will primarily affect the **spectrum** of the programme, with a fast vibrato. Finally above c. 20Hz the result will be most noticeable in **pitch** and **timbre** changes.

Other performers who have used RMs in live performance are Toshi Ichiyanagi, Arne Nordheim, Roger Reynolds, and in this country Richard Orton, Roger Smalley, Tim Souster and myself.

The circuit that follows is the most basic, primitive RM circuit, and is entirely passive, not requiring a power supply. It can be found in various books and articles; the only addition to it here is the suggestion of which comparatively cheap and easily obtainable transformers and diodes should be employed; others are of course also feasible. A number of more sophisticated designs are also available, often transformerless/integrated circuit balanced modulators, mainly in synthesizers. Some of these are based on analogue multiplier computer circuits. Each has advantages and disadvantages. One of the problems in designing any form of RM is coping with 'breakthrough' of one of the inputs. It appears to be impossible to eliminate this leakage entirely. In my experience, however, breakthrough only really occurs when an oscillator is being used as the carrier, and it may be caused by the oscillator signal being present on its earth lead (which cannot be disconnected as it prevents hum). When no oscillator is used, unfortunately a less common situation, even with this basic RM circuit any leakage will be completely suppressed by slight adjustment of the various volume controls that are involved; and in such a case, this simple device, containing no additional circuitry to reduce breakthrough (which also makes the output sound blander), gives a sharper, stronger sound than, for example, the commonly used RMs in the EMS Synthi range (I have directly compared them). This rougher sound can be smoothed if required, but with the blander sound, the reverse is not possible. A good example of this is the part composed by Michael Robinson in Gentle Fire's "Group Composition V", for cello ring-modulated by sounds pre-recorded on a long tape loop; whenever the cello stopped playing, no breakthrough was audible.

When the carrier is an oscillator, it is often possible to cover up much of the leakage in live performances; even with the most expensive professional RMs the breakthrough is still faintly audible. In this connection it would sometimes be valuable to have on the oscillator the possibility to change frequency in a way that avoids the intervening glissandi that normally occur, perhaps by using a linear slide controller or even a keyboard for rough tuning; in performance it is usually too risky to turn down the oscillator's volume control each time its frequency is changed, because of the very delicate balance of the inputs that is necessary to obtain the right quality of modulation. A further possibility for minimising oscillator breakthrough is to insert a preamplifier between it and the RM, in order to **reduce** the voltage of the oscillator output! Even with a poor quality preamplifier this can prove effective, since any distortion it introduces will tend to be lost in the modulated sound.

Components used: two transformers, ratio 1:1 with secondary centre-tapped. – e.g. RS T/T1 transistor transformer **or** Eagle LT-44 transformer (even though



the latter's ratio is 5:1!); four germanium diodes – e.g. OA5, OA70, OA81. Total cost, including case and input/output sockets, £4 – £5. Mount the circuit in a metal box (if built into a studio installation, a metal front panel should suffice). Both inputs and output are high impedance.

Various modifications can be made:

1. if the sources of the two inputs (e.g. tape recorder, microphone preamplifier, oscillator) do not have their own output level controls, it is advisable to place a potentiometer across each RM input, c. 500k log. The same applies to the output.

2. for a more precise balance between the two inputs (thus also reducing breakthrough), instead of Input 2 being directly joined to the two transformer centre-taps, it can go to the wipers of two potentiometers (value to be found by experiment) placed across the secondaries of each transformer; in addition, resistors may be placed between the outer terminals of each potentiometer and the diode ring.

3. the 'direction' of the diode ring is unimportant and may be reversed. Suggestion for experimentation: explore what happens when one or more of the diodes in the ring are reversed (cf. the diode bridge arrangement for power supplies). Also note the interchangeability of inputs and output.

4. it may prove necessary to earth the metal casing of the transformers.

5. to minimise breakthrough it is advisable to connect any oscillator to Input 1; however, due to the fact that the two inputs enter the circuit at different points (unlike the identical inputs of the more sophisticated balanced RMs), different timbre qualities can be obtained by swapping the two inputs; a sound that modulates well with a low-frequency carrier may not be so effective with a high frequency one, unless the inputs are reversed. A 2-pole-2-way switch permits rapid comparison.

6. a switch may also be inserted to bypass the RM entirely, enabling Input 2 to be connected directly to the output. Alternatively, the original programme may be mixed with the output by means of two potentiometers in different amounts, including either one alone.

Additional experiments can be made by connecting an identical signal to both inputs (similar to an 'amplitude filter'); exploiting the breakthrough by only putting a signal into Input 2; splitting the output into two parallel channels and using the extra one via a potentiometer as the signal for one of the inputs; and by cascading two RMs, as in a frequency shifter, with a single oscillator providing the carrier for both, with or without a filter following either or both RMs. With each of these there will be a very different output level from the RM.

Most synthesizer RMs are not obtainable as separate modules. Some Frequency Shifters (but not those mentioned below) are unsuitable, being designed to prevent acoustic feedback and having a range of only c. ± 5Hz. The following individual units are available (prices vary considerably, up to nearly £400 for Buchla Model 285):

- ARP modules for Series 2000 synthesizers (e.g. 2500): 1005 Modamp (balanced modulator, voltage-controlled amplifier); 1035 Triple modulator (three balanced modulators, six microphone preamplifiers). Both need special power supply and patching connections. There is also ARP encapsulated function block (circuitry only) 4014 Balanced Modulator. UK agent F.W.O.Bauch, Borehamwood, Essex.
- Buchla Associates, California (no UK agent): Model 285 Frequency Shifter / Balanced Modulator (presumably contains own power supply and probably supersedes earlier models 111A Dual RM and 185 Frequency Shifter).
- EMS London have just released a separate Phase-Frequency Shifter (own power supply).
- Moog units 6401 RM and 6402 dual RM (own power supplies); 6552 Frequency Shifter (needs special power supply). At the time of writing it is not known if these are still available. UK Agent: Henri Selmer & Co., Braintree, Essex.

Further information regarding synthesizers can be found in the magazine Studio Sound (e.g. special survey, May 1975).

Oberheim Electronics (USA) produce a Music Modulator (used by Don Ellis and Miles Davis among others), which includes a microphone preamplifier, a sine-wave generator as carrier (which can be replaced by an external signal), and a potentiometer which controls the blend of the original programme with the RM output (with sockets to enable the controls for these two to be replaced by footpedals). A similar device, the Maestro, is available in this country (possibly an updated version of the above?).

A cheaper transistorised synthesizer RM is available from Dewtron, Ferndown, Dorset. RM-2 can be obtained as a boxed unit complete with battery power supply or as a basic module.

It is also possible to purchase the components and circuit diagrams for RMs published as synthesizer modules in electronics magazines (for the two versions of the RM used in the PE Sound Synthesizer and the PE Minisonic, published in Practical Electronics in Aug. 1973 and Jan. 1975 respectively, see recent advertisements in the magazine).

Mullard produce an integrated circuit TAB-101 Ring Modulator, and the technical information sheet suggests a circuit using it. The Oberheim RM is based on another IC, the Motorola MC1495 Analog Multiplier. Other more recent ICs can probably also be used as the basis of a RM (see Maplin Electronic Supplies catalogue).

Bibliography

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Editor's note

While much of the information in this note is now purely historical, there appears to be a resurgence of interest in simple ring modulators and many of the practical details remain worth consulting.

This article originally appeared in the London based magazine "MUSICS", No.6, p3-5, February/March 1976. The initial dissemination of MUSICS was not wide and there are few library holdings, so the article has been reissued to allow free dissemination via the internet. It was retyped as the original copy was poorly printed and simply scanning produced unreadable copy. The copyright, film rights etc. remain with the estate of Hugh Davies and, presumably, the editors of MUSICS. This reissue is intended as a note of thanks to Hugh Davies who did much to show the world that electronic music was possible without the resources of institutional electronic music studios.

While ring modulation works by frequency multiplication as described, at least some of the unique sound of the classic diode-transformer ring modulator is caused by saturation and non-linear effects in the transformers and by distortion cased by the non-zero turn-on voltage of the diodes. These effects are determined by the exact choice of component and are not particularly amenable to analysis. The article probably has insufficient emphasis on the choice of diode, in particular the requirement to match germanium diodes (which were manufactured to fairly wide tolerances) in order to minimize carrier breakthrough. It is worth experimenting with Schottky diodes which have a low turn-on voltage and are now widely and cheaply available (e.g. 1N5711, 1N6263 etc.). As the article emphasises, breakthrough is problematic and is considerably worsened by the presence of small DC offsets or low frequency mains pickup in the programme channel. DC offsets cause the diodes to begin to conduct and the carrier bleeds through, while mains pickup modulates to give signals at carrier frequency plus or minus the mains frequency. These are largely indistinguishable from the raw carrier. The description of problems with oscillator outputs and the suggestion of putting a pre-amp between the oscillator and the ring modulator to improve matters suggest to me that Hugh possessed an oscillator with a DC offset or mains hum on its output.

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