A High-Accuracy Frequency Shifter for Professional Audio Applications*

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Shifting of the frequency spectrum of an audio signal is a powerful technique for modifying the qualities of musical sounds. The subject design brings frequency shifter ("Klangumwandler") technology up to current professional audio standards. Performance specifications include 70-dB dynamic range, 40-dB rejection of unwanted sidebands, amount of frequency shift continuously variable through as much as —5000 to +5000 Hz, simultaneous availability of positively and negatively shifted signals, and variable threshold squelch.

INTRODUCTION: The frequency shifter referred to in this paper is not a transposing device but an electronic instrument which is capable of shifting all of the frequencies of the audio spectrum by the same amount. Because of this feature, a sound with a harmonic overtone structure is changed into one with nonharmonic overtones which entirely alters the character of the program material. This instrument, therefore, is a most powerful tool in the hands of the electronic music composer or performer who is searching for new audio processes and musical effects.

Frequency shifters, with varying degrees of detuning capabilities, have been around for a number of years. Basically, antifeedback devices belong in this category. It is known that by shifting speech frequencies by a small amount, say 5 Hz, the acoustical feedback in a PA system is noticeably reduced. The devices developed and built for this purpose are limited to speech frequencies for which they perform well [1], [2].

Another frequency shifter, with limited detuning, has been built as a multiple single-sideband device for electronic organs to simulate the choral tone effect [3].

A frequency shifter capable of larger changes of musical frequencies has become known under the name "Klangumwandler," and has been used in the German broadcasting system [4]. This device operates through double heterodyning and the use of a single-sideband filter.

Another frequency shifter, also capable of large changes of musical frequencies, which operates on the phase-shifting principle for single-sideband production, has become widely known as the Bode frequency shifter.

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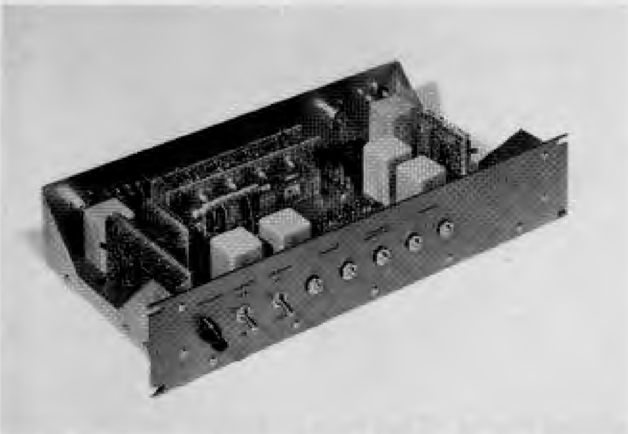


Fig. 1. Bode frequency shifter which operates on phase-shifting principle.

Fig. 1 shows this device in a standard rack mounting configuration. One of the features is a built-in squelch circuit, controlled by the program material, which suppresses the carrier when the program level is below a preset value.

A special frequency shifter, which has been built in limited quantities, combines the features of the heterodyning principle with those of the phase-shifting principle for single-sideband production [5].

A detailed description of the two basic means of frequency shifting follows.

HETERODYNING-TYPE FREQUENCY SHIFTER

In the heterodyning-type frequency shifter the program material is mixed in a balanced modulator with an ultrasonic carrier of, for instance, 20 kHz. Thus two sidebands are generated around a suppressed carrier of 20 kHz. One of these sidebands, preferably the higher one, is passed through a single-sideband filter with a very steep attenuation around the carrier frequency for suppressing the unwanted sideband down to the lowest possible frequency (closest to the carrier). The passed sideband is then mixed in a second balanced modulator with a second ultrasonic frequency around 20 kHz, and thus reconverted into the audio range.

Through this process of double heterodyning and single-sideband filtering, the program material is shifted by an amount equal to the difference between the two ultrasonic frequencies. For instance, when using a carrier frequency of 20 kHz, an upper sideband filter, and a second ultrasonic frequency of 19.9 kHz, the reheterodyned program material appears shifted up by the difference of the two ultrasonic frequencies, or by 100 Hz.

The quality of the heterodyning-type frequency shifter depends mainly upon the quality of the single-sideband filter. The upper limit frequency of shifters built with known types of single-sideband filters lies at 10 kHz. The lower frequency limit for no interference by the suppressed sideband is in the order of 200 Hz if the performance of the instrument is not enhanced by special means, such as the combination of the heterodyning and phase shifter methods for single-sideband production [5].

PHASE-SHIFTING-TYPE FREQUENCY SHIFTER

The operation of the phase-shifting-type frequency shifter is different from that of the heterodyning type. For basic understanding of the operating principle a simplified schematic block diagram as shown in Fig. 2 is helpful. Here the program material is fed into the input f_1IN and to two phase shifting circuits α_1 and β_1, which are so designed that at their outputs of all the frequencies of the audio range are 90° apart in phase. These output

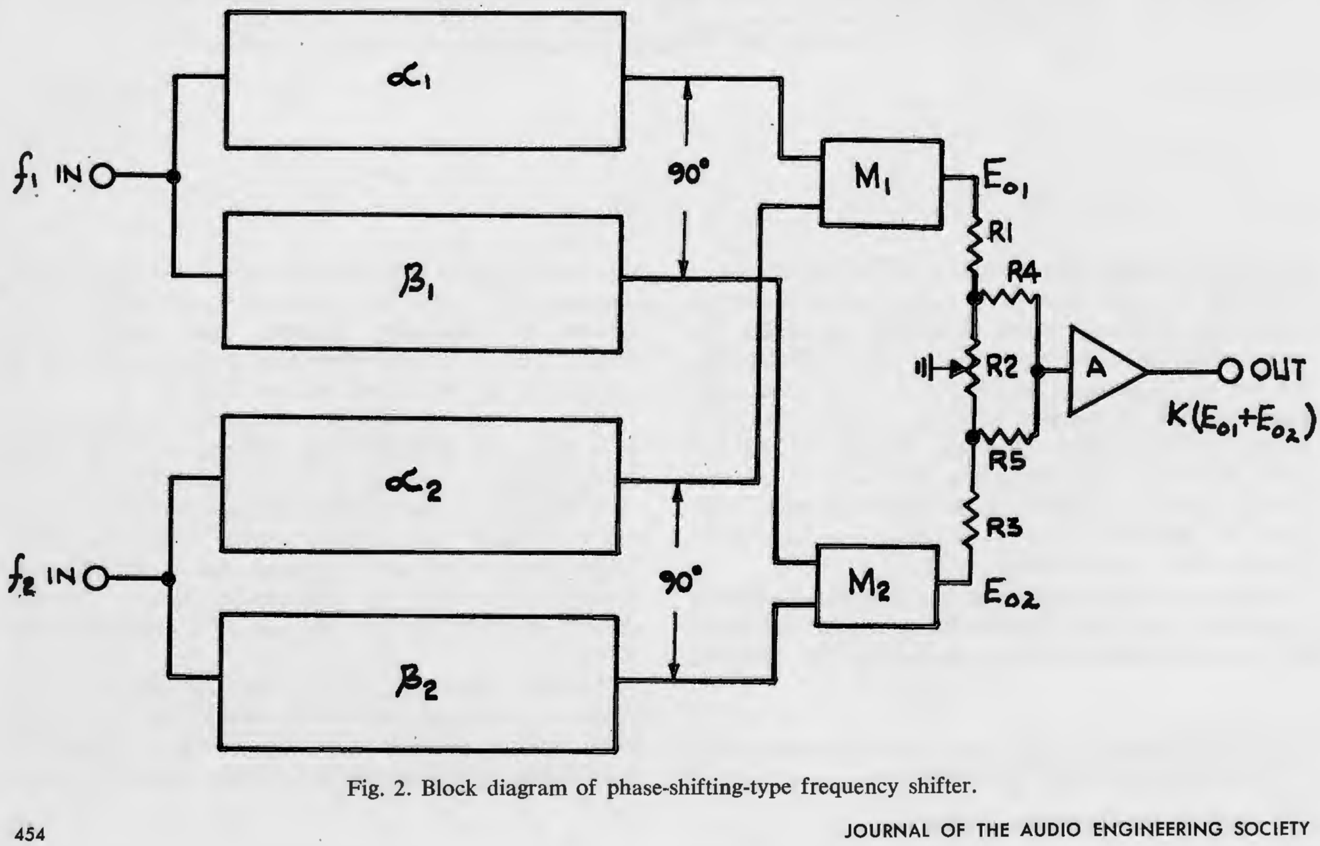


Fig. 2. Block diagram of phase-shifting-type frequency shifter.

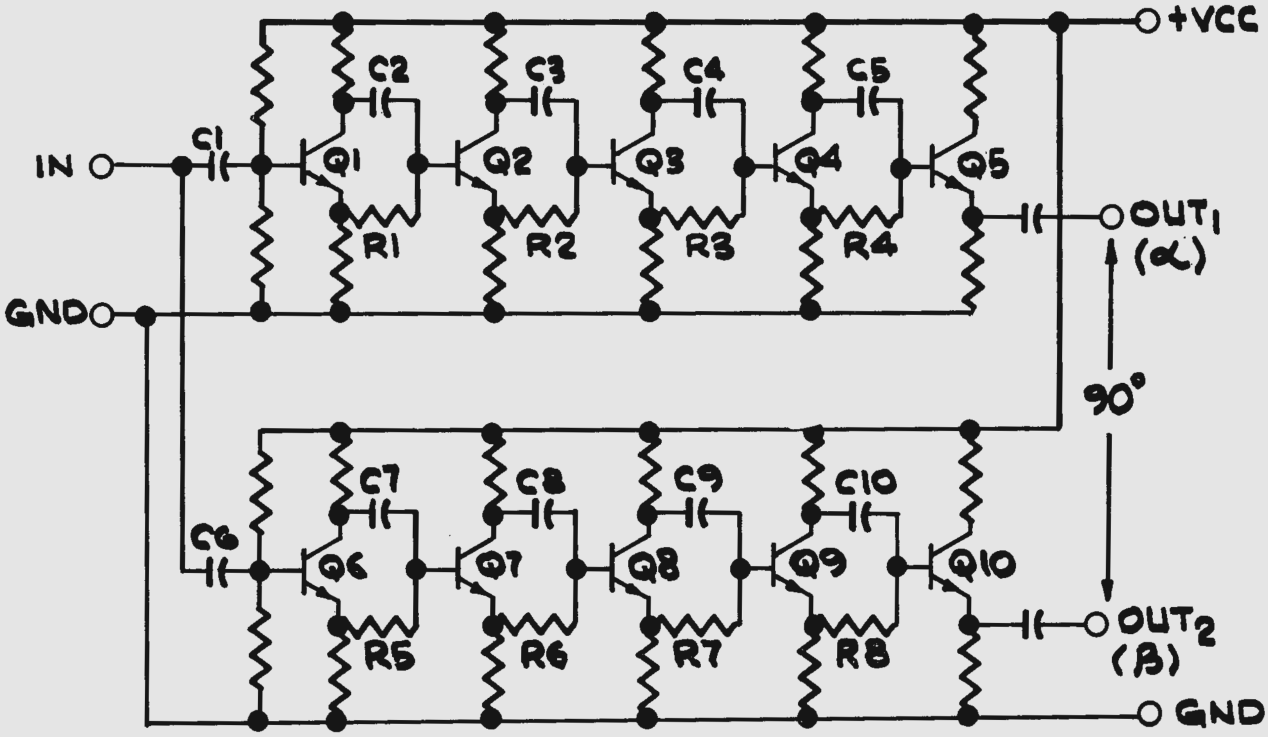


Fig. 3. Dome filter for producing 90° phase difference.

frequencies are then fed to two multipliers M_1 and M_2.

In the same way the carrier frequency, which is fed to f_2IN and the phase shifters α_2 and β_2, is processed for obtaining a relative phase relationship of 90° of carrier frequencies down to about 40 Hz at the outputs of these two phase shifters. These output frequencies are fed into the second inputs of the multipliers M_1 and M_2.

A schematic diagram of one pair of these phase-shifting circuits is shown in Fig. 3. These phase shifters, also referred to as dome filters, are comprised of a number of phase-shifting sections with "center" frequencies equally spread over the audio range in such a way that the cumulative phase shift of one branch is always 90° different from that of the other branch over the specified audio range.

The wave patterns shown in Fig. 4 illustrate in some detail what happens in a frequency shifter according to Fig. 2. Assuming that Fig. 4**a** represents a program frequency and Fig. 4**b** a carrier frequency (by the amount of which Fig. 4**a** is to be changed), then Fig. 4**c** represents the waveform at the output of one of the multipliers which receives waves **a** and **b** at its inputs. Since the inputs of the other multiplier receive two waves which are 90° out of phase relative to **a** and **b**, the resulting product will be the waveforms shown in Fig. 4**d**. By summing outputs **c** and **d**, the waveform according to Fig. 4**e** results, the frequency of which (in the case of this example) is that of **a** + **b**.

FREQUENCY SHIFTER OF NEW DESIGN

The new frequency shifter also utilizes the phase shifting principle. While the frequency shifter just described is limited to accept carrier frequencies between 40 Hz and 10 kHz, the new instrument is designed for frequency shifts to and through zero. When going through zero, the sign of the obtained sideband is changed, which is to say that an up-detuning becomes a down-detuning and vice versa. Through the full utilization of the inherent possibilities of the system employed, two complementary sidebands are generated, one of which represents a frequency shift in a direction opposite to the other. The performance of this new highly accurate instrument is realized through

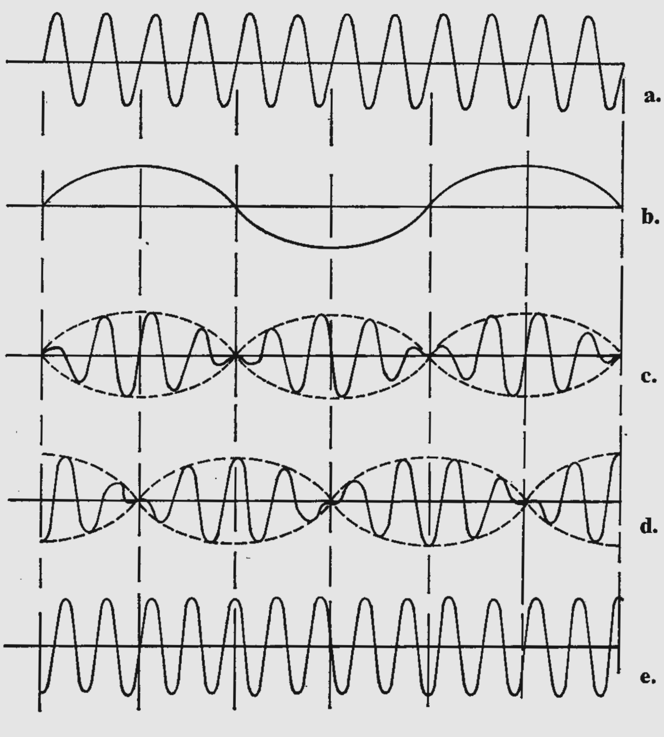


Fig. 4. Typical wave patterns in phase-shifting-type frequency shifter. **a.** Program input. **b.** Carrier input. **c, d.** Outputs of two multipliers. **e.** Sum of **c** + **d**. The frequency of **e** is the arithmetic sum of the frequencies of **a** + **b**.

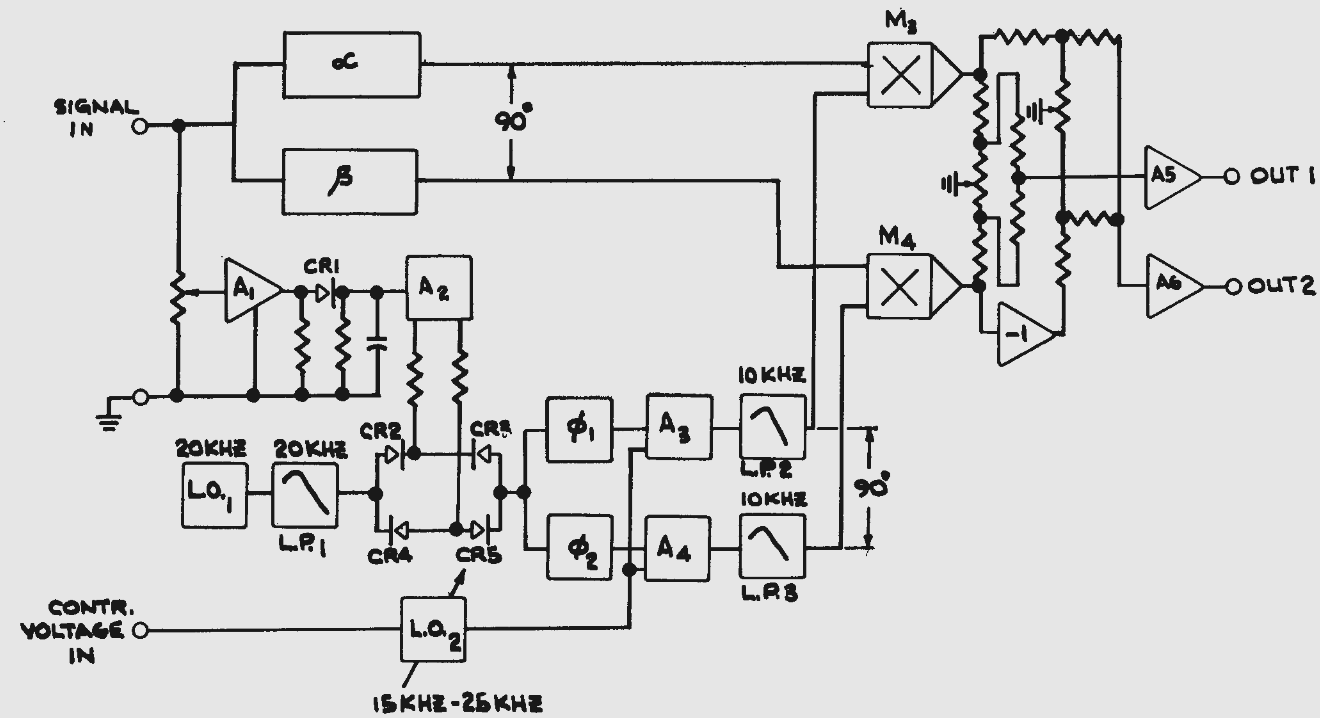


Fig. 5. Block diagram of new system.

the utilization of the latest state-of-the-art circuitry.

Fig. 5 shows a schematic block diagram of the new system. The audio signal is fed to the input of the phase filter pair comprising sections α and β. The 90° out-of-phase outputs of these phase filters are then fed to the first inputs of the multipliers M_3 and M_4.

The second inputs of M_3 and M_4 receive two 90° out-of-phase components of a sine wave which is generated in a beat frequency oscillator consisting of one fixed 20-kHz oscillator (LO_1) and one voltage-controlled 15–25-kHz oscillator (LO_2). The output of LO_1 is passed through a low-pass filter to obtain a pure sine wave, and through a squelch gate, consisting of diodes CR_2 through CR_5, which is switched by amplifier A_1, rectifier CR_1, and bias supply circuit A_2. This gate passes the oscillator frequency in the presence of a program level above a presettable threshold.

The frequency of LO_1 is subsequently fed to two phase shifters ϕ_1 and ϕ_2, which yield two outputs 90° out of phase. In the multiplier A_3 and A_4 these 90° out-of-phase frequencies are mixed with the variable frequency of LO_2, and then low-pass filtered in LP_2 and LP_3. The signals at the outputs of these filters are the beat frequency with two 90° out-of-phase components.

This principle of obtaining beat frequencies in 90° phase relationship works extremely well. Fig. 6 shows two typical circular oscilloscope patterns obtained by feeding the components into the X and Y inputs at a beat frequency of 1 kHz and 1 Hz, respectively.

As mentioned before, these beat frequencies are fed into the second inputs of M_3 and M_4 and after summing of the output signals, a frequency (or frequency spectrum) results, which is the sum or the difference of the input (program) frequency and the beat frequency.

The second frequency shift is obtained by adding the output of M_3 to the inverted output of M_4. With this arrangement, the complementary sideband appears at OUT_2, and the frequency shift is therefore equal and opposite to that of the signal at OUT_1.

In addition to the two complementary frequency-shifted program signals obtained at the outputs OUT_1 and OUT_2, a ring modulator product can be derived from any of the outputs of M_3 or M_4.

An important feature of this new system is the inclusion of a stable voltage-controlled oscillator LO_2. Accurately programmable frequency shifts, including shifts around and through zero, are generated by applying control voltages that are directly proportional to the magnitude of the desired frequency shifts. In addition, a temperature-stable exponential function generator permits the use of a pattern of control voltage differences to program corresponding frequency shift ratios.

The frequency range of the dome filter pair in this instrument has been extended to 16 kHz for a minimum of 40-dB suppression of the unwanted sideband, as compared with 10 kHz on previous models.

SOME TYPICAL APPLICATIONS OF NEW FREQUENCY SHIFTER

The new frequency shifter can be programmed with suitable synthesizer or audio system modules to obtain a wide variety of new sounds and musical effects. A few typical applications are as follows.

Programming with Periodic and Aperiodic Voltage Functions

1) By setting the main tuning control to zero and applying a low-frequency square wave to the control voltage input in the linear mode, the up and down detuned outputs will switch places, resulting in a new stereo-type effect when heard over two stereo channels. The character of this effect is dependent upon the square-wave frequency (typically, for instance, 5–6 Hz) and its amplitude. Entirely different effects are achieved when raising this

frequency above 20 Hz. The amplitude of the square wave will determine the amount of detuning for both sidebands, which will become attractive when the detuning frequency is in some harmonic relationship to the fundamental of a simple program material.

2) By setting the main tuning control to zero beat and applying a vibrato-type sine wave to the control voltage input in the linear mode, a vibrato-type effect is created, displaying the widest frequency shift at low program frequencies and decreasing toward higher frequencies.

3) With the main tuning control in the zero beat position, the application of a sawtooth wave to the control voltage input in the linear mode produces a dramatic effect when the sawtooth frequency is at about 1–2 Hz and when the main tuning knob is slowly turned out of its center position.

4) By setting the main tuning control to zero and applying a pink noise with limited voltage to the control voltage input in the linear mode, the program material assumes a hoarse quality which can be remixed with the original sound.

Programming with Predetermined Detuning Sequences

To program the frequency shifter with a predetermined sequence of detuning frequencies, these frequencies can be prerecorded on a tape and converted to a corresponding sequence of control voltages. These are then fed into the control-voltage input set to the exponential mode. Thus the control voltages will be retransformed into the original frequency sequence, which will determine the frequency shifts of the two outputs, which will be complementary to each other.

Programming of Detuning Frequencies with a Sequencer

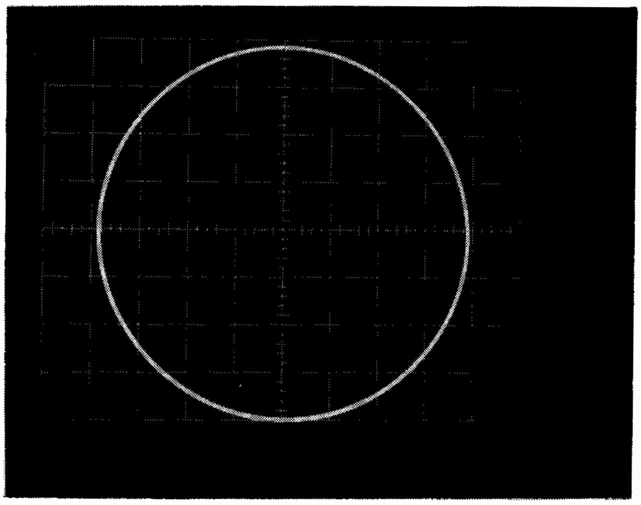
For a periodic recycling of the detuning (or frequency-shift) increments a sequencer can be employed. The control voltages of the sequencer can be fed to the control voltage input of the frequency shifter either in the linear or the exponential mode, depending upon the type of effect desired.

Multichannel Stereo Effects

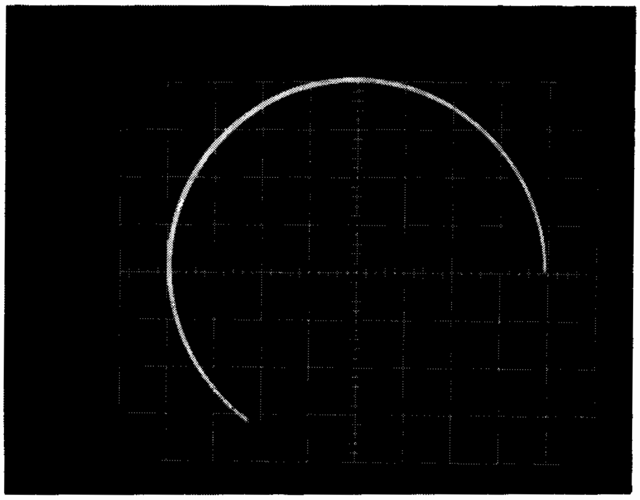
A four-channel stereo effect can be achieved by feeding the direct program material to channel 1, the voltages of the output OUT_1 to channel 2, the voltages of the output OUT_2 to channel 3, and the ring modulator product to channel 4. This effect can be further enhanced by employing some of the sequencing or modulation techniques just described.

Modulation with Complex Sounds

Many areas of experimentation are possible by feeding complex sounds into the control input and varying the character of the program material from simple to complex sounds, including the use of the same program material for the program and the control inputs.



a.



b.

Fig. 6. Oscilloscope patterns showing 90° phase difference between two outputs of beat frequency oscillator. Exposure times about ½ second. **a.** 1 kHz. **b.** 1 Hz.

Iteration Effects

The program material (preferably a sequence of percussive sounds) is fed through a mixer to the recording head of a tape delay device. A playback head feeds back the information to the frequency shifter, one output of which feeds it to a second input of the mixer. The shifted sound is rerecorded, played back, and frequency shifted repetitively, so that an increasingly detuned sound is created, the character of which is determined by the amount of tape delay and the amount and sign of detuning.

A simulated iteration effect can be produced by connecting a sequencer to the control voltage input of the frequency shifter and setting the eight control voltages so as to simulate successive up- or down-detuning. The effect is especially attractive when the program material is synchronized with the sequencer. In this setup both outputs of the shifter can be used, with one creating an up-detuning effect and the other a down-detuning effect.

These are but a few examples of the possible applications of the new frequency shifter and the variety of new sounds and musical effects obtainable with it.

APPENDIX

BASIC EQUATIONS FOR THE PHASE-SHIFTING-TYPE FREQUENCY SHIFTER

Assume that the audio signal frequency at the output of phase filter a_1 of Fig. 2 is

$$E_{a_1} = A \cos \omega_1 t \tag{1}$$

and that the carrier frequency at the output of phase filter a_2 is

$$E_{a_2} = B \cos \omega_2 t \tag{2}$$

and that, for the sake of simplification, $A = 1$ and $B = 1$. Then after multiplication in the four-quadrant multiplier M_1 (assuming a scale factor of 1) the output of M_1 will be

$$E_{o_1} = \cos \omega_1 t \cos \omega_2 t \tag{3}$$

or, after transforming the equation,

$$E_{o_1} = \tfrac{1}{2} [\cos (\omega_2 + \omega_1) t + \cos (\omega_2 - \omega_1) t]. \tag{4}$$

Due to the 90° phase relationship between the outputs of phase filter β_1 versus phase filter a_1, and phase filter β_2 versus phase filter a_2, the output of β_1 (in the case of the polarity chosen for this example) will be

$$E_{\beta_1} = C \cos (\omega_1 t + \phi) \tag{5}$$

or for $\phi = 90°$,

$$E_{\beta_1} = C \sin \omega_1 t. \tag{6}$$

In the same way, the output of β_2 is

$$E_{\beta_2} = D \cos (\omega_2 t + \phi) \tag{7}$$

or for $\phi = 90°$,

$$E_{\beta_2} = D \sin \omega_2 t. \tag{8}$$

Again, assuming $C = 1$ and $D = 1$ and a scale factor of 1 for the four-quadrant multiplier M_2, the output of M_2 will be

$$E_{o_2} = \sin \omega_1 t \sin \omega_2 t \tag{9}$$

or, after transforming the equation,

$$E_{o_2} = \tfrac{1}{2} [\cos(\omega_2 - \omega_1)t - \cos(\omega_2 + \omega_1)t]. \tag{10}$$

The summing of the outputs of M_1 and M_2 can be expressed by summing the Eqs. (4) and (10), or

$$\begin{aligned} E_{o_1} + E_{o_2} = {} & \tfrac{1}{2} \cos(\omega_2 + \omega_1)t + \tfrac{1}{2} \cos(\omega_2 - \omega_1)t \\ & + \tfrac{1}{2} \cos(\omega_2 - \omega_1)t - \tfrac{1}{2} \cos(\omega_2 + \omega_1)t \end{aligned} \tag{11}$$

in which the expression $\tfrac{1}{2} \cos(\omega_2 + \omega_1)t$ is canceled so that

$$E_{o_1} + E_{o_2} = \cos (\omega_2 - \omega_1) t \tag{12}$$

which is the expression of a single-sideband frequency, or a frequency shift of ω_2 by a negative amount of ω_1.

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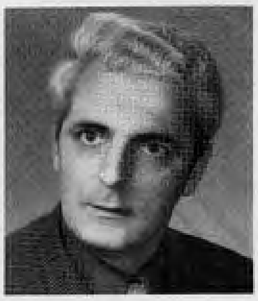
THE AUTHORS



Harald Bode, a native of Hamburg, Germany, graduated from the University of Hamburg in 1934. He continued his education at the Heinrich Hertz Institute of Berlin. After the Institute, Mr. Bode specialized in the field of electronic music, creating the Bode-Melochord and pioneering the industrial production of electronic organs (Polychord) in Germany. Coming to the United States in 1954, he held several positions before joining the Bell Aerospace Company in Niagara Falls, New York, where he is currently specializing in the field of microelectronics, while remaining active in the field of electronic music instrument design. Bode has written a variety of German and American publications on audio and electronic music and holds more than 20 patents in the United States, Canada and several European countries.

Robert A. Moog holds a PhD. in Engineering Physics from Cornell University. He has been actively engaged in electronic music instrument design since 1954. At the present time, he is president of Moog Music, Inc., manufacturer of Moog Synthesizers and related audio signal generating and processing instruments. Moog is a Fellow of the Audio Engineering Society, and a member of the Institute of Electrical and Electronic Engineers and of the Acoustical Society of America. Recent awards include the National Academy of Recording Arts and Sciences Trustees Award (1970), the Billboard Trendsetter Award (1970), and the New York State Small Businessman of the Year Award (1970). He has authored many articles on electronic music instrumentation and holds one patent in the field.