presented at the 61st Convention for the
Audio Engineering Society
November 3-6, 1978
New York

Appears in the AES Journal
Volume 27, Number 9 pp. 673 (1979)

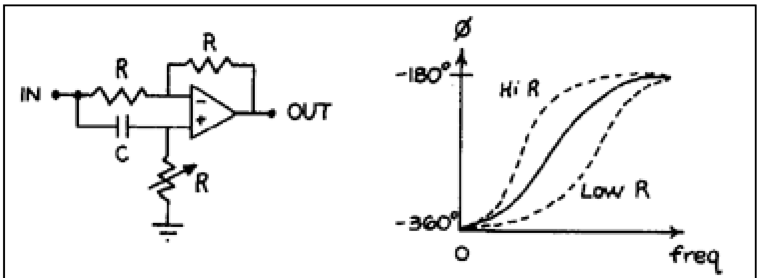
A DIGITAL "PHASE SHIFTER" FOR MUSICAL APPLICATIONS, USING THE BELL LABS (ALLES-FISCHER) DIGITAL FILTER MODULE

by Michael Beigel

ABSTRACT
A programmable digital filter bank is configured to simulate a swept analog "phase shifter." The operation of the Bell Labs Digital Filter Module and its use in simulating the swept filter response are described. Analog end digital models for phase-shifters are compared, and directions for further development are indicated.

INTRODUCTION
Swept phase-shifting devices have become a widely-used and effective means of modifying the tonal and spatial characteristics of musical signals. In anticipation of all-digital musical and audio-processing systems in the near future, the phase-shifting effect has been implemented using the Digital Filter Module of the Bell Labs Digital Synthesis System. The digital method of "phase-shifting" has its own set of peculiarities, advantages, and disadvantages, and these will be discussed. Digital filter models and appropriate coefficients are presented for use in other digital processing systems.

ANALOG PHASE-SHIFTER
The analog phase-shift system is already well known in the electronic music field, and will be summarized here as a background for the digital implementation. For a more detailed treatment of the mathematical aspects of analog phase-shifters, see Reference 1.



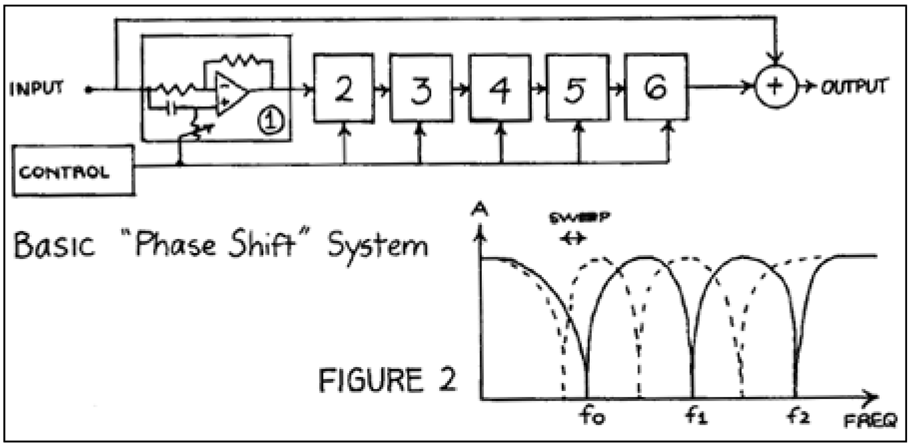
$$H(s) = \frac{s-a}{s+a} \qquad a = \frac{1}{RC}$$

$$\emptyset = (-180° - 2\tan^{-1}\omega RC)$$

"Phase-Shift" Stage

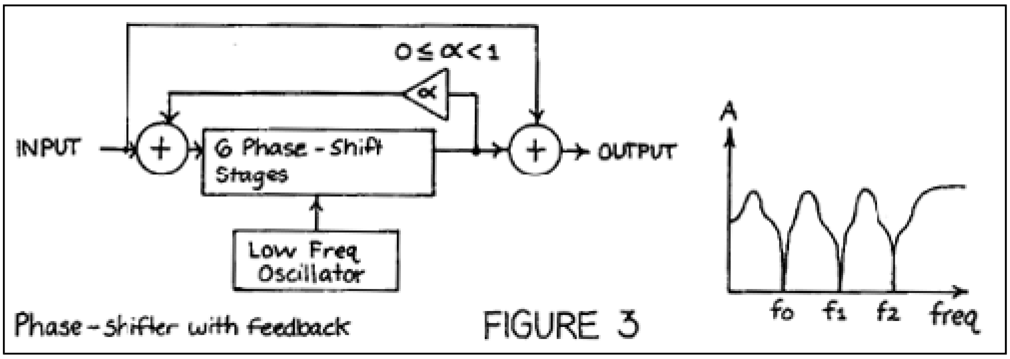
FIGURE 1

Originally used as a substitute for a variable time-delay in musical vibrato and "rotating sound" systems, the variable phase-shift element is a unity-gain amplifier which provides a phase-shift of -360 degrees at low frequencies, approaching -180 degrees at higher frequencies. A popular implementation of the single phase-shift stage, its transfer functions and phase-frequency response are shown in Figure I (See also Reference 2).

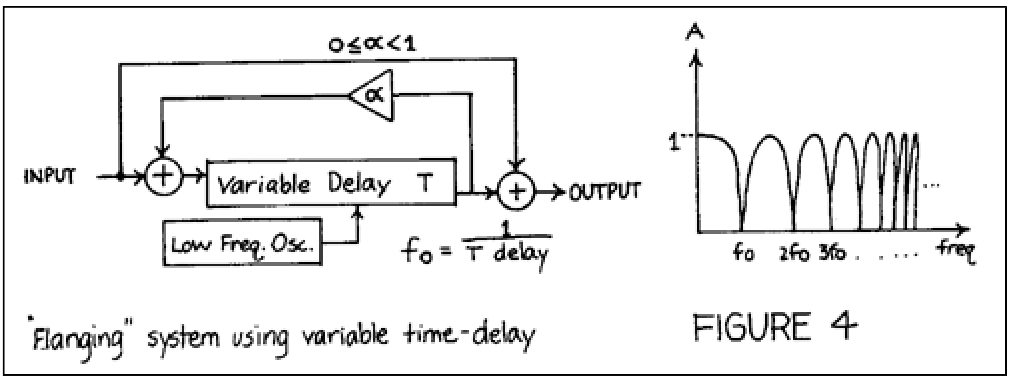


These phase-shift stages are cascaded (usually in multiples of 2) and the final output is summed with the input signal, resulting in cancellations in the frequency response. For six phase-shift stages with identical time-constants the frequency response and system diagram is shown in Figure 2.

An enhancement of this response is obtained by using positive feedback around the phase-shift sections, whereby resonant peaks as well as cancellations in the frequency response are obtained. Phase shifters are usually swept by a low-frequency sine- or transverse-wave oscillator, and the added resonances of the feedback-enhanced system provide greater definition of the effect. See Figure 3.



DIGITAL PHASE-SHIFT MODEL
The most obvious digital approximation to a phase-shifting system is a variable-length digital delay. The audible effect of such a system, however, is recognizably distinct from the phase-shifting system previously described, and is not always preferable. A class of sampled analog or digital delay systems has been implemented with the product designation of "flangers". The system diagram and frequency response (see Figure 4) reveal a large number of harmonically related peaks and cancellations which can overshadow the musical signal in many applications.

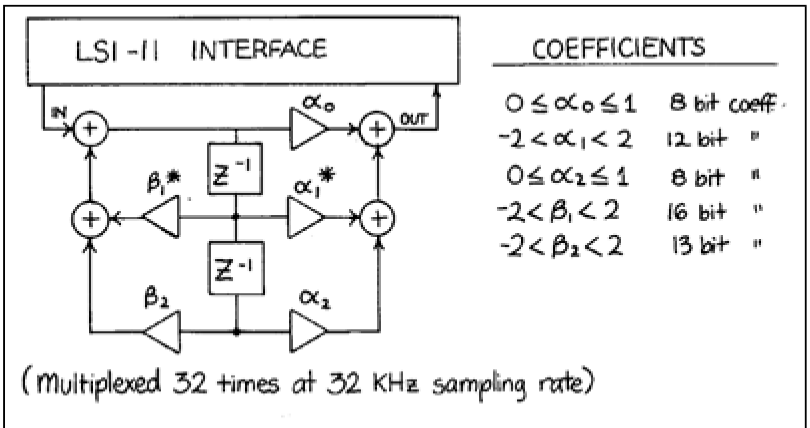


In the present case, it was decided to simulate the characteristics of the six-stage analog phase shifter, using the Digital Filter Module available on the Bell Labs Digital Synthesis System. Digital filter stages approximating the transfer function of the phase-shift stage (Figure 1) are cascaded and connected in the same configuration as described in Figure 3. using digital filter sections and envelope-multipliers available on the Bell Labs Digital Synthesizer.

USING THE BELL LABS DIGITAL FILTER MODULE

The Bell Labs Digital Filter Module is a general purpose system which provides signal-processing functions for a wide range of applications in telecommunications systems, as well as high-quality audio signal processing. For this application, many of its specialized functions are not used, and a simplified model of the filter system is presented for the sake of clarity

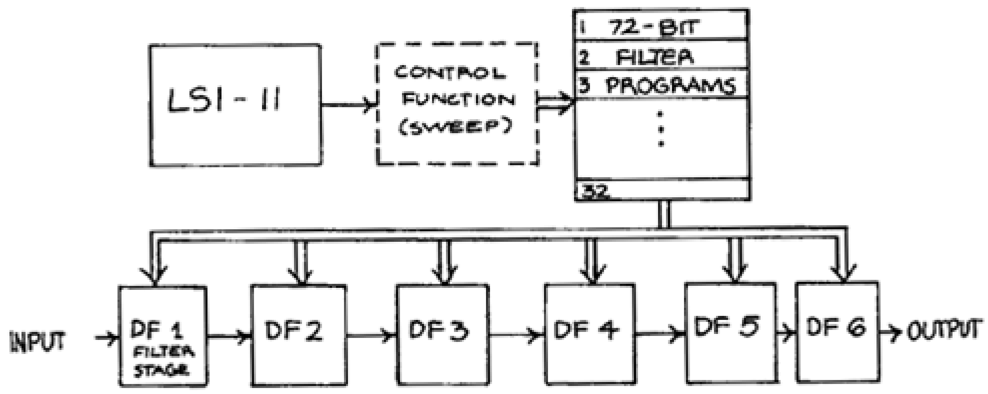
The filter system is a time-multiplexed real-time processor which provides thirty-two second-order filter stages operating at a 30 kHz sampling rate. The module interfaces to an LSI-II computer by means of a IK-word address space in the LSI-II memory. By means of this interface, the module is configured in terms of the filter coefficients, interconnection of filter stages, and other control signals. A simplified diagram of the basic filter stage is shown in Figure 5. Overload protect, precision rectification, full-scale limiting, and specialized interconnect functions are available; a more complete specification may be found in Reference 3.



DIGITAL FILTER SECTION-SIMPLIFIED FIGURE 5

Each of the filter sections is completely configured by a 72-bit program. The program specifies the filter coefficients (a0, a1, a2, B1, B2) and a number of other processing options. Thirty-two of these programs are available, and are used here for specifying phase-shift stages with different time-constants. In order to provide a filter with dynamically variable properties, two methods are available. In the first case, coefficients a1 and B1 may be directly specified using special registers. The coefficients can either be stored in the LSI-ll's main memory or computed directly using an appropriate algorithm.

In the second case, one of the thirty-two stored filter programs may be selected by an appropriate control function: each separate filter stage can then have thirty-two possible values. The advantage of this second method is that all of the filter coefficients (A0...B2) are accessible and very little real-time computation is involved (See Figure 6). A disadvantage is the limited number of available programs, resulting in a possible audible "quantization" of swept filter effects. However, satisfactory results were obtained using the second method of filter sweeping for the phase-shift model.

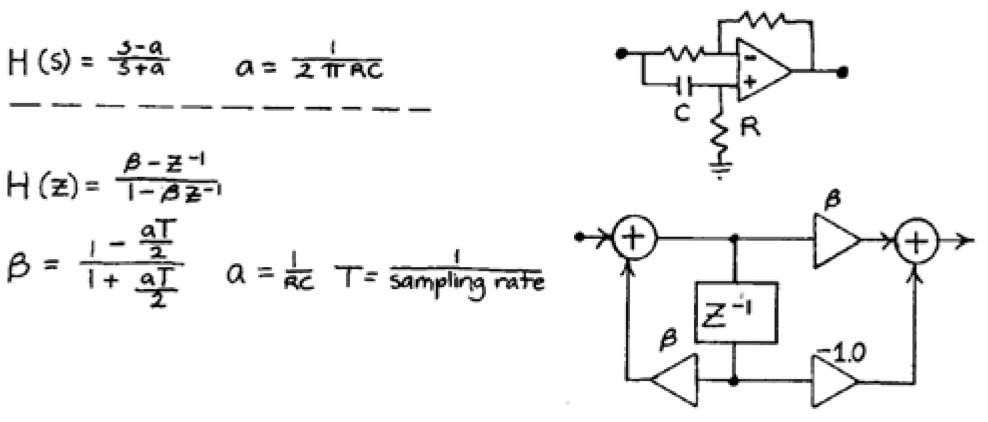


Program - selecting sweep function FIGURE 6

The individual filter stages may be interconnected and cascaded in a wide variety of configurations. Used as a sub-section of the Bell Labs Digital Synthesizer, the filter programs and interconnections are specified by means of a special operating system designed for the synthesizer by Doug Bayer (See Reference 4). The operating system enables the user to specify all programs and configurations in a high-level language (Bell Labs "C" compiler) with minimum attention to hardware-oriented details, and integrates the filter module into the totality of the Digital Synthesis System.

DIGITAL PHASE-SHIFT FILTER STAGES

To implement the Phase-Shift transfer function for the digital filter model it is necessary to convert the analog transfer function into an equivalent z-transform function, and to derive coefficients equivalent to the analog filter's parameters over the range of interest. For a single phase-shift stage, the analog and digital equivalent systems are given (Figure 7):



One- Stage Model FIGURE 7

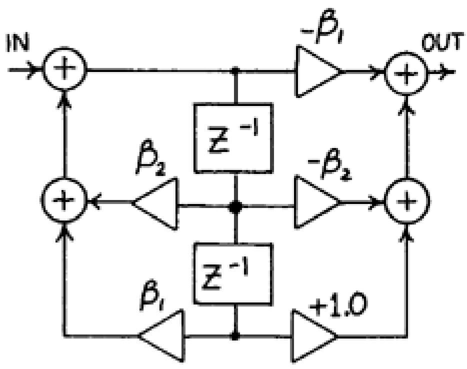
This configuration uses only one-half of the available processing in filter section , so another digital filter simulating two analog stages was constructed (Figure 8):

$$H(s) = \left(\frac{s-a}{s+a}\right)^2 = \frac{s^2-2as+a^2}{s^2+2as+a^2}$$

$$H(z) = \frac{-\beta_1 - \beta_2 z^{-1} + z^{-2}}{1 - \beta_2 z^{-1} - \beta_1 z^{-2}}$$

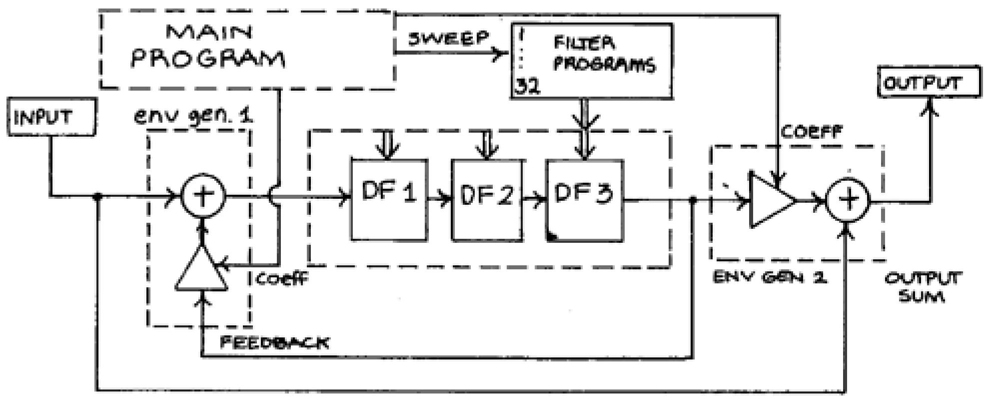
$$\beta_1 = -\frac{(\alpha-1)^2}{(\alpha+1)^2} \qquad \beta_2 = \frac{2(\alpha-1)}{(\alpha+1)}$$

$$\alpha = \frac{2}{aT} \qquad a = \frac{1}{RC}$$



Two - Stage Model FIGURE 8

CONFIGURING FOR "SWEPT PHASE SHIFTER" SYSTEM
The phase-shift stages, as described, are connected in a manner analogous to the swept phase-shifter with feedback (Figure 3), using the unit generators available in the Bell Labs Synthesizer. Phase-shifters employing six "single" filter stages or three "dual" stages have been constructed (Figure 9).



Digital "Phase Shifter" System FIGURE 9

The envelope-generators ENVI and ENV2 are each composed of a program-controllable digital multiplier (with coefficients between zero and 1) and a summing junction (the multiplier coefficient is 1) for the input signal and the phased signal, and can also function as a switch or mixer for the phased signal. ENV2 is controlled by a front-panel slider or a program variable, and controls the amount of positive feedback around the phasing loop.

The thirty-two filter programs contain coefficients to sweep the phase-shift response over an approximate 5-octave range, sufficient for nearly all processing applications. The sweeping operation is accomplished by incrementing or decrementing the filter program numbers according to a specific sweep function. If all filter stages are incremented at once, the sweep range is divided into 32 gradations of approximately .156 octave each. For finer resolution of sweep, the filter stages can be incremented one at a time, resulting in 96 gradations of .052 octave for the "double" filter sections, or 192 gradations of .026 octave for the "single" filter sections. This provides a smooth-sounding sweep as required for any specific application.

The use of direct computation of filter coefficients is not implemented here. For the present filter stages the required coefficients are not all accessible, and the models would have to be changed.

COMPARISON OF ANALOG AND DIGITAL PHASE SHIFTERS

The digital system compares favorably with the best analog systems in terms of sweep accuracy and versatility. Since virtually all analog units are swept over non-critical limits, no attempt has been made to provide devices with a precise relationship between control signals and frequency response. The digital system provides a control function which is inherently as accurate as might be needed. This feature encourages the use of phase-shift filters in novel signal-processing applications.

The 30kHz sampling rate of the digital system, and the associated rolloff of the anti-aliasing filters in the 14-bit ADC system presently used, provide a slightly compromised frequency-response characteristic. Rolloff begins at about 13 kHz, and thus the extreme high-end of the audible frequency response is attenuated. Analog systems can currently provide better high-frequency performance.

The dynamic range of the present system, which uses 16-bit binary arithmetic, is also slightly inferior to the best available analog systems. The theoretical dynamic range is about 90 dB using internally generated 16-bit signals, or 78 dB using signals from the 14-bit ADC in the Bell Labs Synthesizer. Two other factors further limit these values: the use of positive feedback around the phasing loop will reduce the effective dynamic range in proportion to the overall feedback gain, and the results of rounding errors in the arithmetic will further reduce the signal-to-noise ratio. The results of the arithmetic (rounding) errors increase at low input signal frequencies, and also for the lower values of frequency response. In these cases, the ratio of change in the input values to the limits of the arithmetic resolution becomes significant, and instability can result. Therefore the dynamic range becomes quite compressed under these conditions. The rounding and scaling errors are proportionally worse for the two-stage filter sections than for the one-stage filters, since second order calculations are involved in the former case.

MUSICAL APPLICATIONS, AND DIRECTIONS FOR FURTHER RESEARCH

The conventional notion of the Phase-shifting effect is that of a periodic or "rotating" effect which lends interest and motion to musical signals. Less commonly used variations are envelope or pedal-controlled sweep of the filters to provide multi-pole dynamic filter effects, or multiple channel phasing to provide spatial vibrato effects.

The accuracy and versatility of the digital system, especially in the context of a programmable multi-channel synthesizer, introduces additional possibilities. Among these are:

1). Multi-pole filter effects in which the filters emphasize definite harmonics of the musical signal
2). Precisely controllable spatial vibrato and quadraphonic rotation effects
3). Stepped or "arpeggiated" sweep patterns to produce overtone "harmonics" within individual notes or chords.

The additional programmable control of the mixing and feedback functions provides a facility for dynamic variations in the intensity or prominence of the effect. Many combinations and extensions of these techniques should provide materials for creative new uses of "phase-shifting."

In order to make the digital approach more viable in terms of frequency response and dynamic range, two hardware modifications are indicated:

1). Sampling rate could be increased to 40 kHz or more, at the expense of the number of available filter stages. At present, a quadraphonic system of four six-stage filters uses at most 24 of the 32 available filter sections on the Bell Labs Module.
2). Internal arithmetic could be expanded beyond the present 16-bit architecture. To provide excellent 16-bit "terminal" characteristics, an architecture of 24 bits binary or 18 bits floating point is indicated.

ACKNOWLEDGMENTS

This research was made possible by the generous assistance and guidance of the following people: H. G. Alles, Greg Sims, Doug Bayer and J. F. Kaiser of Bell Telephone Laboratories at Murray Hill, N. J.; and R. L. Kosut of Singer Company, Binghamton, N. Y.

REFERENCES

W. M. Hartmann, "Flanging and Phaser's"
J. Audio Eng. Soc., vol. 26, pp 439-443, (June 1978)

Jerald G. Graeme, Applications of Operational Amplifiers
pp 102-103. McGraw-Hill Book Co. c 1973

H. G. Alles and W. C. Fischer, "A New General Purpose Digital Filter"
October 7, 1976, Bell Telephone Laboratories

Douglas L. Bayer, "Real-Time Software for a Digital Music Synthesizer"
Computer Music Journal Volume I, Number 4

Back to Articles

Home | RFID Group | Services | Project Profiles | Technologists | Business Terms | Product Development | Articles | Contact | Links

Phone : (760) 633 - 3868 ... Fax : (760) 633 - 3819