Shaped Tone-Burst Testing

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A properly shaped tone burst is used to evaluate the dynamic behavior of a loudspeaker within narrow frequency bands. The raised-cosine envelope of a five-cycle burst reduces the low-frequency content of the test signal and confines the spectrum to a one-third-octave width.

The transient behavior of the loudspeaker is indicated by a change in the envelope of the burst signal. The frequency response of the loudspeaker is related to the maximum amplitude of the received burst. The relatively short duration of the burst preserves time domain information and gives a slightly smoothed frequency response.

Discrimination against echoes is obtained from the short duration of the shaped tone burst. The influence of room reflections on the measurement is minimized.

1 MEASUREMENT OBJECTIVES

The test technique described here developed out of the need for evaluating and characterizing loudspeaker drivers under free-field conditions.

The usual techniques of swept measurements in an anechoic chamber, pulse testing, and associated Fourier transform or time-delay spectrometry were unavailable or too costly for the application [1], [2].

Tone-burst testing has been used for a long time to evaluate the performance of loudspeakers [3]. Often the response of the loudspeaker to the turn-on and turn-off of the burst signal is observed and taken as an indication of the loudspeaker's ability to follow transients. If the burst is sufficiently long in duration, so that turn-on transients have time to decay, then the amplitude of the burst becomes a measure of the steady-state frequency response.

To perform free-field measurements in a reflective environment, data have to be taken before the first echo arrives at the measurement location. The duration of the tone burst, therefore, has to be short, or the received burst has to be gated to discriminate against echoes [4]. In both cases the measurement moves toward the beginning of the burst and is therefore more subject to turn-on transients.

Tone-burst testing looks attractive because of the simplicity of generating the test signal and measuring selected portions of the envelope. Considerable care has to be taken, though, in interpreting the results [5]–[7]. A tone burst of frequency f_0 and burst length T in the time domain is characterized by a spectrum with maximum energy centered at frequency f_0 and of a width proportional to $1/T$, and with significant extension to higher and lower frequencies. The spectrum rolls off toward low frequencies at a very slow rate, which is immediately apparent from a spectrum display on a logarithmic frequency scale (Fig. 1).

This broadband characteristic of the tone-burst spectrum can cause considerable envelope distortion due to gain and phase variations of the network under test at frequencies below f_0. To illustrate this effect, a five-cycle burst is applied to a second-order high-pass filter of $Q_p = 2$ at pole frequency f_p (Fig. 2). The burst signal is viewed on a linear amplitude scale (bottom traces) and on a logarithmic scale after full-wave rectification (top traces). Compared to the input signal [Fig. 2(a)], the response to the burst at frequency f_p clearly shows the slow turn-on and extended turn-off transients of the filter resonance [Fig. 2(b)]. Excitation of the network at $3f_p$, for example, shows the envelope distortion caused by the resonance below the burst frequency and the corresponding turn-off transient at one-third the excitation frequency. Clearly, even with a gating technique, the relevant information for the response at $3f_p$ would be difficult to select [Fig. 2(c)].

Easy interpretation of the test data and especially clear identification of localized resonances were a major objective for the measurement technique under investigation.

2 SHAPED TONE BURST

The envelope of the tone burst can be modified to reduce the low-frequency spectral content. This modification is similar to the premultiplication of a signal with a window function when performing a discrete Fourier transform analysis. All the different window functions developed for this technique to contain the spectral content apply also to the tone-burst shaping [8]. The usual "tone burst" is merely a continuous sine wave multiplied with a rectangular window. The Hanning window or raised-cosine envelope $w(t) = ½ - ½\cos 2\pi t/T$ has been found adequate for the shaped tone burst (Fig. 3).

From a network analysis viewpoint the following steps take place (Fig. 4): A continuous sine wave $s(t)$ is multiplied with the window function $w(t)$ to generate the shaped burst $x(t)$ which is then applied to the network under test. The output $y(t)$ from the network with impulse response $h(t)$ is the convolution between the input $x(t)$ and $h(t)$. The output signal $y(t)$ can be observed on an oscilloscope. Amplitude and envelope will differ in general from $x(t)$, except when the impulse response $h(t)$ is much shorter in duration (including the tail) than $x(t)$. Following the same steps in the frequency domain, the single frequency signal f_0 is convolved with the spectrum of the window, thereby translating it from zero frequency to f_0.

The output spectrum $Y(f)$ is then merely the product of the frequency response $H(f)$ of the network as it would have been measured with a continuous sine wave signal and the shaped burst spectrum $X(f)$.

The output frequency response $Y(f)$ will approach the steady-state frequency response $H(f)$ as the window $w(t)$ is lengthened, since the input spectrum width is inversely proportional to the burst length T and approaches that of a continuous sine wave f_0.

The burst length chosen must be a compromise between meaningful frequency response information $Y(f)$ and useful transient information from the envelope of $y(t)$.

Only the time function $y(t)$ is easily observed and analyzed on an oscilloscope. The peak amplitude of this signal is taken as the magnitude of the frequency response $Y(f)$. It can be seen empirically that if the burst builds up and decays very gradually so that the network can follow in every detail, then the peak amplitude reached will equal the steady-state frequency response $H(f)$. Likewise as the burst is shortened, the amplitude of $y(t)$ becomes a function of the network response speed and an indication of how accurately the network will follow a band-limited transient signal. The duration T was chosen for a main-lobe width of the burst spectrum of approximately a third of an octave. This is in recognition of the psychoacoustic observation that for narrower bandwidths the spectral distribution of sound does not influence loudness. Therefore one-third-octave resolution for $Y(f)$ seems adequate. This then leads to a five-cycle shaped tone burst. The number of cycles is constant, and the window time length decreases with increasing frequency, thus holding the spectrum width to a constant percentage of the center frequency f_0.

The relatively short burst is advantageous in discrimination against echoes, which furthermore improves with increasing frequency as the window length decreases. The low-frequency limit for free-field measurements is reached

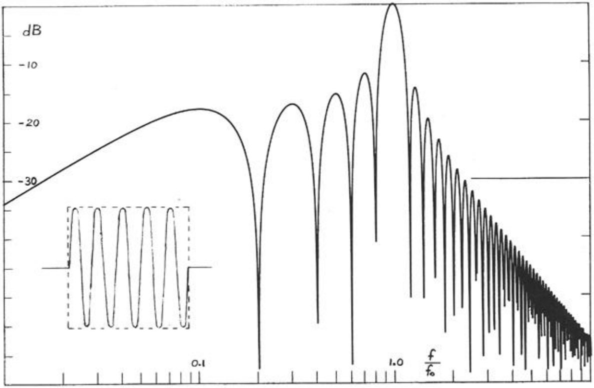

Fig. 1. Spectrum of a five-cycle tone burst of frequency f_0. The high-frequency rolloff approaches 12 dB/octave, but only 6 dB/octave for the low frequencies.

when the difference in propagation time between the direct signal and the first reflected signal equals the burst length.

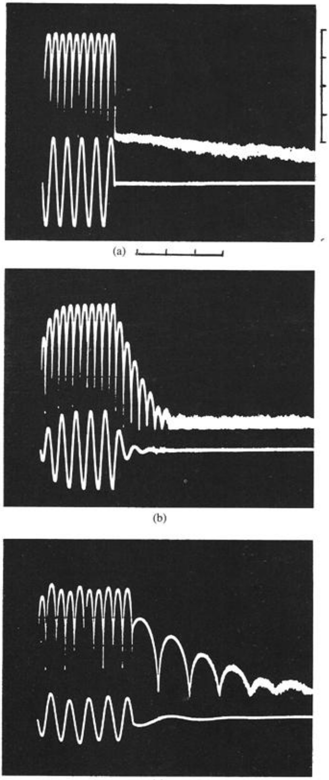

Fig. 2. Time response of a second-order high-pass filter ($f_p = 1$ kHz, $Q_p = 2$) to a five-cycle tone burst. Top traces show the full-wave rectified burst response on a logarithmic amplitude scale of 10 dB/div. Note the enhanced visibility of the burst envelope on the log scale compared to the linear amplitude scale for the bottom traces. (a) 1-kHz input signal to filter (2 ms/div). (b) Response at 1 kHz with slow decay due to 1 kHz resonance of filter (2 ms/div). (c) 3-kHz input tone burst (0.5 ms/div). Burst envelope and decay are strongly modified by the 1-kHz resonance which is excited by the low-frequency content of the 3-kHz tone burst.

Below this frequency, echoes can introduce errors depending on their relative amplitude and phase. Room dimensions, the proximity to reflecting surfaces, and their absorption characteristics affect the shaped tone-burst test in the same way as they affect other techniques [1], [2], [4]. Echoes, though, are easily recognized on an oscilloscope because of the distinctive shape of the burst.

3 MEASUREMENTS

The frequency response of the high-pass filter, which was investigated using a rectangular burst in Fig. 2, was measured using a continuous sine wave and a shaped tone burst as test signals (Fig. 5). There is close agreement between the two measurements, except for a slight broadening of the peak in the network response due to the limited resolution of the burst containing a one-third-octave spectral content instead of a single spectral line as for the sine wave.

In the time domain the envelope distortion is clearly visible at the network resonance frequency, but well suppressed at three times that frequency (Fig. 6). The advantage of the shaped burst over the rectangular tone burst is clearly visible from a comparison of Figs. 2(c) and 6(c). Not only is the lower frequency decay transient considerably attenuated, but also the peak amplitude is clearly defined for the shaped burst. It is not obvious where to measure the amplitude of the rectangular burst.

Loudspeaker drivers often show high Q resonances involving cone and surround interaction. These are easily identified by the envelope distortion of the shaped burst response, even if nothing unusual is seen in the frequency response curve (Fig. 7). Thus the limited resolution in the frequency domain is overcome by additional time domain information. Localized high Q resonances are difficult to pinpoint in swept frequency magnitude response curves and often require careful analysis of the associated phase response. The shaped burst, in contrast, gives a meaningful description of the network dynamics within narrow frequency bands. The peak amplitude reached is the steady-state frequency response modified by the speed with which the network can follow the input. The length of the output burst is an indication of the stored and slowly released energy due to resonances. The shaped burst concentrates the energy into a narrow frequency band, and transient information is therefore frequency selective and not as easily masked by system peculiarities at another frequency.

4 TEST SYSTEM

Different techniques could be used to generate the shaped tone burst. In what seems the simplest approach, a signal generator would be 100% amplitude modulated by a sine wave at one-fifth the output frequency of the generator. After each cycle of the modulating waveform, the output signal would be blanked for several hundred milliseconds to allow echoes from the test environment to decay sufficiently. This requires two phase-locked sources and some gating circuitry. A much simpler approach was taken which approximates the raised cosine envelope of the burst by only changing the amplitude of the sine wave every

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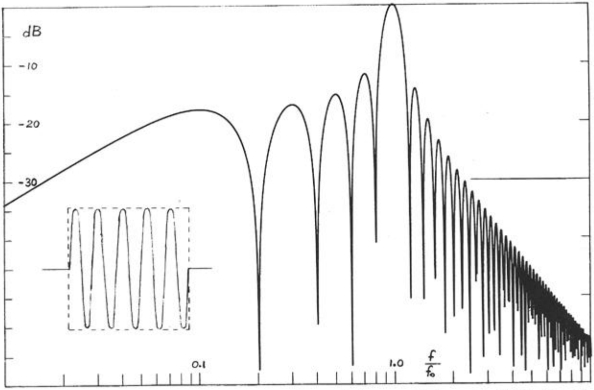

Fig. 1. Spectrum of a five-cycle tone burst of frequency f_0. The high-frequency rolloff approaches 12 dB/octave, but only 6 dB/octave for the low frequencies.

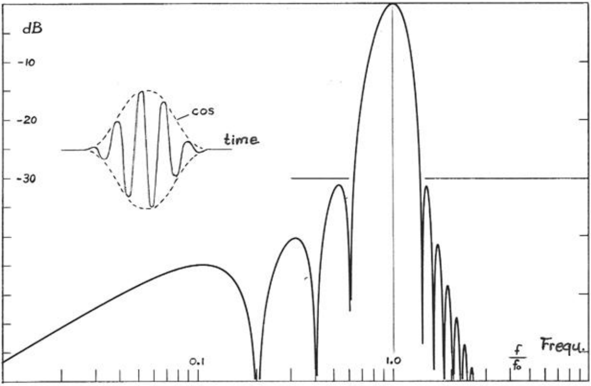

Fig. 3. Spectrum of a five-cycle tone burst with raised-cosine envelope (Hanning window). Main lobe of spectrum is widened compared to a burst with rectangular envelope, but side lobes fall off more rapidly.

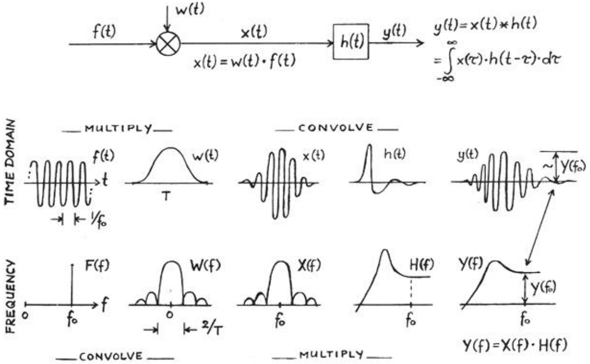

Fig. 4. A shaped burst $x(t)$ is generated by multiplying a continuous sine wave $f(t)$ with a window $w(t)$. The network response $y(t)$ is the convolution of the network impulse response $h(t)$ with the burst signal $x(t)$. In the frequency domain the network response $Y(f)$ is the product of the steady-state sine wave frequency response $H(f)$ and the burst spectrum $X(f)$.

half-cycle. A programmable attenuator is switched at each zero crossing of the sine wave. The slope discontinuity of the burst signal at the zero crossings generates odd-order harmonics. These limit the dynamic range when measuring the low-frequency cutoff attenuation of a high-pass such as shown in Fig. 5. The attenuation coefficients can be optimized for suppression of the low-frequency content of the signal, resulting in the spectrum of Fig. 8. It compares favorably with the ideal Hanning spectrum (Fig. 3). The test system then consists of a conventional sine wave generator which is modulated and gated with a programmed attenuator (Figs. 9 and 10). The amplitude of the received burst is measured with a peak detector after full-wave rectification. The rectified signal is logarithmically amplified and displayed on an oscilloscope. This logarithmic display enhances the detectability of envelope distortion as exponential decays become straight-lined and the visual amplitude range is increased (Figs. 2, 6, and 7).

5 SUMMARY

The shaped tone-burst meets the requirements for free-field measurements of loudspeakers in closed rooms. It provides a short-duration signal which is analyzed before echoes arrive. The raised-cosine envelope of the burst and constant five-cycle duration concentrate the spectral energy of the burst in a one-third-octave band. Therefore a one-third-octave analysis of the anechoic frequency and time response is obtained. The envelope of the burst allows identification of localized resonances and is minimally affected by out-of-band network behavior; thus experimental data are easy to interpret. The design of a novel signal generator uses greatly simplified circuitry to approximate the desired test signal. Compared to other approaches to free-field measurements, the shaped tone-burst test represents a very powerful solution by combining time and frequency domain information with economy of equipment. It has been found that the frequency response of loudspeakers measured with the shaped tone burst correlates closely to subjective listening impressions. The shaped burst signal appears to be a very sensitive indicator of phase distortion in subjective evaluations of all-pass networks [9], [10]. The low duty cycle of the test signal is also convenient for high-power testing of loudspeakers, especially tweeters, in order to determine their excursion limits and the onset of audible distortion. The influence of cabinet diffraction effects within one-third-octave bands is reflected in the frequency response. Baffle changes in the order of 1 cm can be resolved in the amplitude response at higher frequencies. The shaped tone-burst test has proven

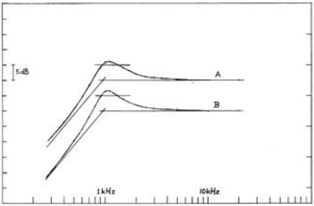

Fig. 5. Frequency response of a second-order high-pass filter (f_p = 1 kHz, Q_p = 2). A—peak amplitude of a five-cycle raised-cosine envelope tone burst of varying frequency after passing through filter; B—swept-frequency or steady-state response. Curves are offset 10 dB. The shaped burst measurement agrees closely with the swept measurement except for some smoothing of the response peak.

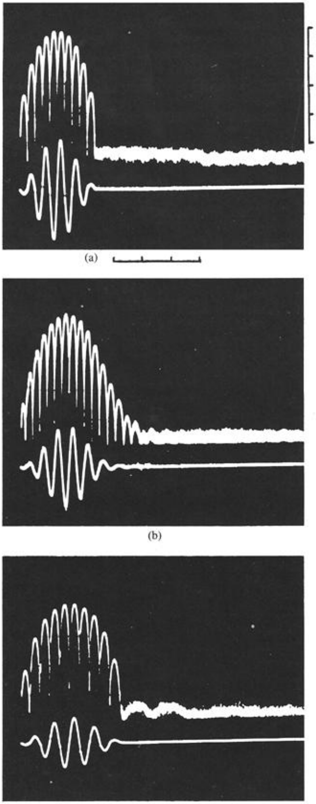

Fig. 6. Time response of a second-order high-pass filter (f_p = 1 kHz, Q_p = 2) to a five-cycle shaped tone burst. (a) 1-kHz input signal to filter. (b) Response at 1 kHz with slow decay due to 1-kHz resonance of filter. (c) Response to a 3-kHz shaped tone burst showing the suppression of the lower frequency resonance. Same scales as Fig. 2.

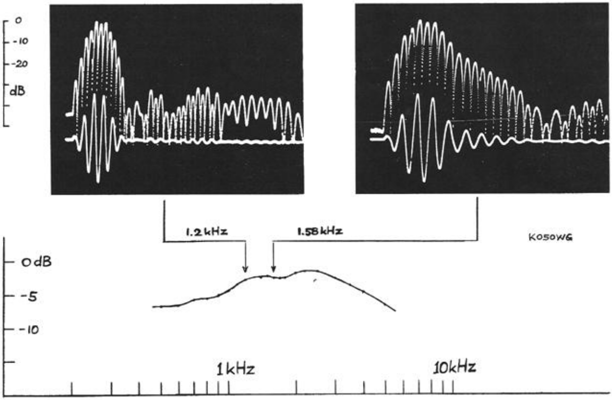

Fig. 7. Shaped tone-burst response of a midrange driver. A 1.58-kHz cone resonance is clearly visible in the burst envelope but not apparent from the burst amplitude response. The resonance is localized and not indicated in the burst response at 1.2 kHz.

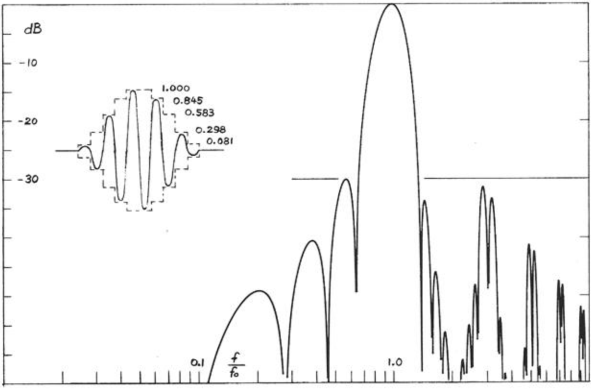

Fig. 8. Spectrum of a five-cycle tone burst which is assembled from half sine waves of different amplitudes to approximate a raised-cosine envelope. The discrete slope changes at zero crossings produce odd-order harmonics. The spectral content compares favorably with the exact raised-cosine envelope spectrum and is an improvement over the rectangular envelope (Figs. 1 and 3).

to be an extremely useful and reliable tool in the development of loudspeaker systems.

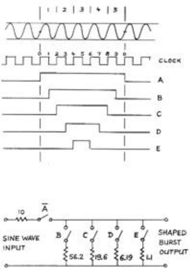

Fig. 9. Component values and timing sequence for a programmable attenuator to generate a tone burst with approximately raised-cosine envelope.

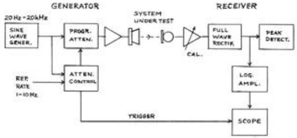

Fig. 10. Block diagram of test system. The shaped tone burst is approximated with a programmed attenuator. Peak measurements are taken by adjusting the calibrated attenuator until the light-emitting diode at the comparator output starts flashing. The burst envelope is observed on an oscilloscope after full-wave rectification and logarithmic amplification.

APPENDIX

In the design of the test system electronics the attempt was made to minimize the component count and to develop simple circuitry which is easily duplicated. This proved to be quite a challenge for the attenuator control circuit which started out with a fair number of flip-flops and gates. The persistence of our colleague, Alan Kafton, led to the circuit diagram for the generator section of the test system shown in Fig. 11.

Measurements have to be taken point by point for the different test frequencies. The calibrated attenuator of the receiver is adjusted until the light from the comparator light-emitting diode just starts flashing. The attenuator reading is manually plotted versus frequency. The resolution and repeatability of amplitude measurements is within 0.2 dB and far exceeds the typical one-third-octave measurement accuracies using a pink noise source.

1) *Generator:* The programmable attenuator portion of the generator is composed of an input buffer amplifier and an output amplifier whose gain is digitally controlled. The gain of the output amplifier is varied by changing the paralleled combination of input resistors. As shown in the timing diagram of Fig. 9, the switches are opened or closed one element at a time, that is, only one switch will change state at each zero crossing. This scheme minimizes switching transients.

The attenuator control portion of the generator is composed of the following integrated-circuit functions:

a) A voltage comparator (LM393) used to amplify and limit the input into a CMOS compatible square wave.

b) A quad EX–OR gate (MC14070). Section A is used to generate a 2-μs pulse at each zero crossing, section B as an inverter, sections C and D for delay.

c) A divide by ten counter (MM74090) wired for biquinary operation.

d) A dual monostable (MC14538). Section A is used as a repetition rate control and section B as a flip-flop to synchronize the start of the burst to a zero crossing.

e) A shift register (MC14035) used to program the attenuator. This circuit is the heart of the attenuator control. When pin 7 is low and a clock pulse appears at pin 6, a one is left shifted into the leftmost bit of the register's 4-bit work (XXXX becomes 1XXX). Likewise, when pin 7 is high, a zero is right shifted into the rightmost bit (1XXX becomes XXX0). In this manner, four consecutive shift-lefts will yield a 1111 output and four consecutive shift-rights will yield a 0000 output. A 0000 output is also achieved by pulling the reset line high.

During the dwell time (burst off) both the counter and the shift registers are reset to zero. When the repetition rate monostable times out, its $\overline{Q}$ output goes high, permitting the sync monostable. At the next positive-going zero crossing (transition 0 in Fig. 9), the monostable turns on section A of the switch and enables the counter and shift register to advance on future zero crossing pulses generated by the EX–OR gate.

On zero crossings 1 through 4 the counter advances, and the shift register shifts left, increasing the burst amplitude. On crossing 5 the counter's Q_A output goes high, but because of its delay, the shift register is again shifted left. On each subsequent crossing, 6 to 10, the counter advances and the shift register shifts right, reducing the burst amplitude. At crossing 10 the Q_A output goes low, triggering the repetition rate monostable, shutting off the burst.

2) *Receiver:* A low-noise input stage with a minimum gain of 20 dB is followed by a calibrated attenuator with a range of ± 10 dB (Fig. 12). The dB scale has nearly uniform spacing. The signal is full-wave rectified in a precision rectifier circuit after another amplification stage with switchable 10 or 20 dB of gain. The light-emitting diode in the comparator and pulse-stretching circuit is turned on when the peak voltage of the rectified signal reaches -1.2 V. The rectified signal is also logarithmically amplified for display on an oscilloscope. The burst generator output signal is applied via a switch to the rectifier and comparator to set its amplitude to 2 V peak to peak by adjusting the signal generator input to the burst generator. This avoids overdriving the circuit and maximizes the ratio of signal to internal switching transients. The switch is also convenient for turning off the shaped tone burst to reduce fatigue when testing loudspeakers. Aside from program material, the

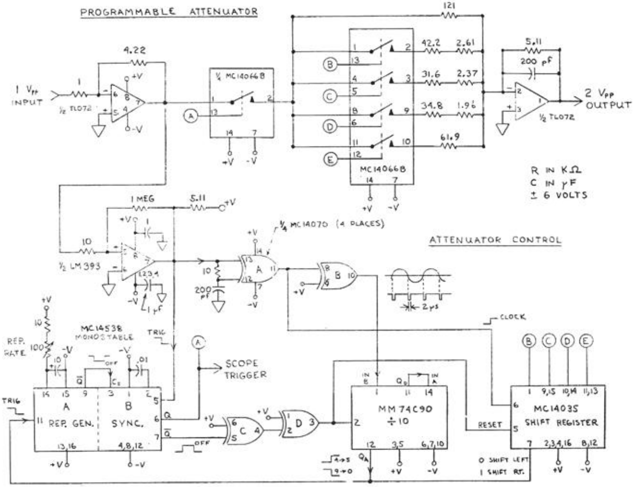

Fig. 11. Shaped tone-burst generator.

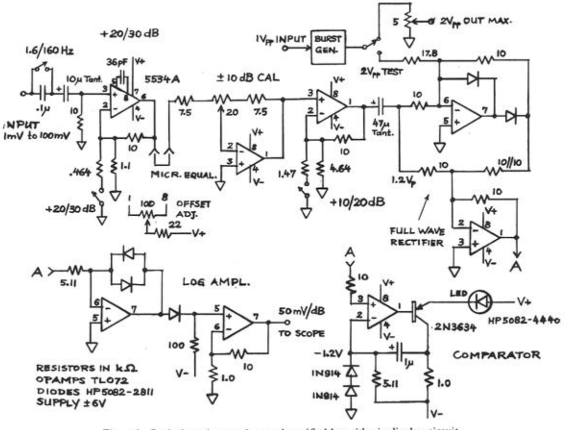

Fig. 12. Peak-detecting receiver and rectified logarithmic display circuit.

shaped tone burst is the most pleasant sounding, and least tiring, test signal known to the author.

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Siegfried H. Linkwitz was born in Bad Oeynhausen, Germany, in 1935. He received the Diplom-Ingenieur degree in electrical engineering from Darmstadt University in 1961. Upon graduation he joined Hewlett Packard Company in California as a research engineer for the design of microwave test equipment. He was involved with the development of signal generators, phase measuring RF voltmeters, and spectrum analyzers. At present Mr. Linkwitz is section manager for spectrum analyzer development.

His interest in audio was stimulated by the gap between reproduced sound and life sound, and he focused on loudspeakers as the least developed component in the reproducing chain.