Dynamic/Transient Intermodulation Distortion (DIM/TIM)

Many audio workers have felt that some audible distortion mechanisms were triggered by changing program material, but not evident with steady-state stimulus such as a single sine wave. In particular, amplifiers with high amounts of negative feedback were singled out as a possible source of such problems due to the time delay inherent in negative feedback loops. The theory was that when a rapidly-changing signal was fed to such an amplifier, a finite time was required for the correction signal to travel back through the feedback loop to the input stage and that the amplifier could be distorting seriously during this time. Dynamic and transient intermodulation test techniques were developed in an effort to isolate these phenomena.

Most proposed DIM/TIM test techniques therefore implement a signal with a rapidly-changing (high slew rate) component. The most popular technique was proposed by Schrock and Otala. The signal (see Figure 29) consists of a band-limited square wave (typically around 3 kHz) plus a high frequency sine wave “probe tone” of one-fourth the peak-to-peak amplitude of the square wave. The rise and fall sections of the square wave stress any portions of the

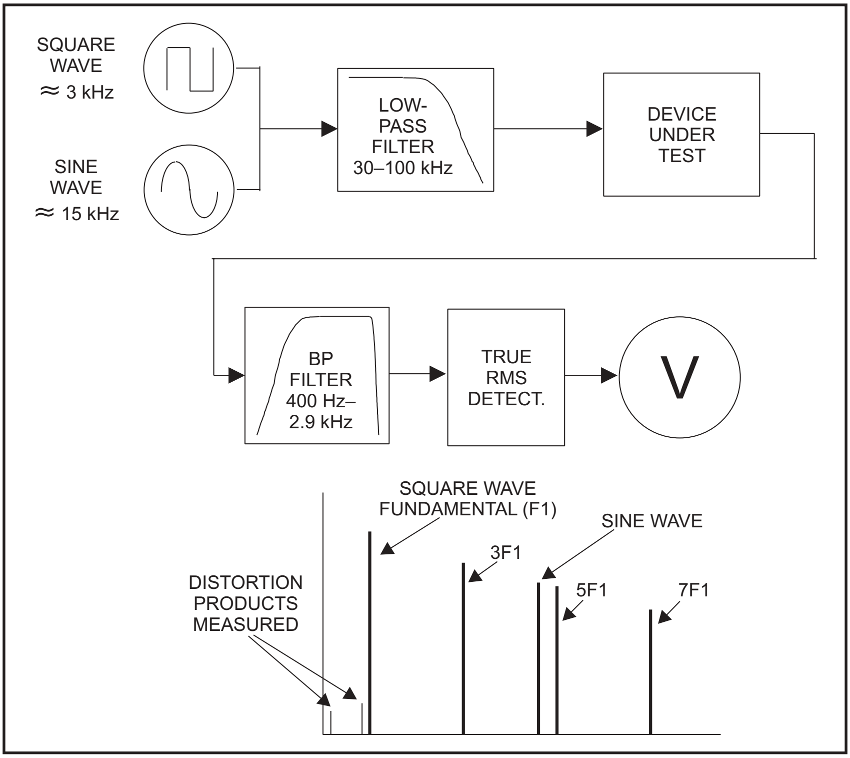

Figure 30. Block diagram, simplified DIM/TIM analyzer.

circuit which have slew rate limitations, and intermodulation products of the sine wave with the fundamental and various harmonics of the square wave will indicate problems. Fully examining the spectrum of such a signal requires an FFT analyzer or other selective spectrum analyzer and a degree of skill. Skritek proposed a simplification in which, by proper selection of sine wave frequency, both an even order and an odd order IMD product would fall into the mid-frequency spectral region below the fundamental of the square wave. This simplified technique can be rather simply implemented with a sophisticated bandpass filter followed by an rms detector, and provides a “one number” readout which does not require operator skill to interpret. Figure 30 is a block diagram of this instrument architecture.

DIM/TIM Instrument Criteria

The square wave generator portion of the oscillator must be followed by selectable low-pass filtering, since both 30 kHz and 100 kHz band limited versions of the signal are proposed by Otala. The stress which the test signal places on the device under test is principally determined by this band limit value. The analyzer must have an extremely well-designed bandpass filter (particularly the lowpass section) in order to reject the square wave fundamental component while providing a low residual distortion value for IMD products falling only a few hundred Hz below the square wave fundamental. Two versions of the technique have been proposed; one for relatively non-band-limited devices such as power amplifiers, and one for band-limited devices or systems such as 15 kHz broadcast links. In order to comply with both versions, the square wave frequency must be settable to either 3.15 kHz or 2.96 kHz and the sine wave probe tone to either 15 kHz or 14 kHz. The detector must be true rms since it is typically measuring a complex signal consisting of at least two components at different frequencies plus noise.

Other IMD Techniques

Several other techniques have been proposed for IMD measurement by workers including Cordell and Thiele. Some use three sine waves; some use a sine wave plus a ramp (triangle) waveform. Few have yet been designed into commercial instrumentation. There are also other IMD measurements which are used in audio-related fields such as telecommunications, but not in consumer audio, pro audio, or broadcasting.

Frequency Measurements

Most modern audio test sets include frequency measurement capability. This capability is almost invariably implemented by standard frequency counter architecture shown as diagrams A and B in Figure 31. In A, a gate is opened for a precise period of time determined by a quartz clock and cycles of

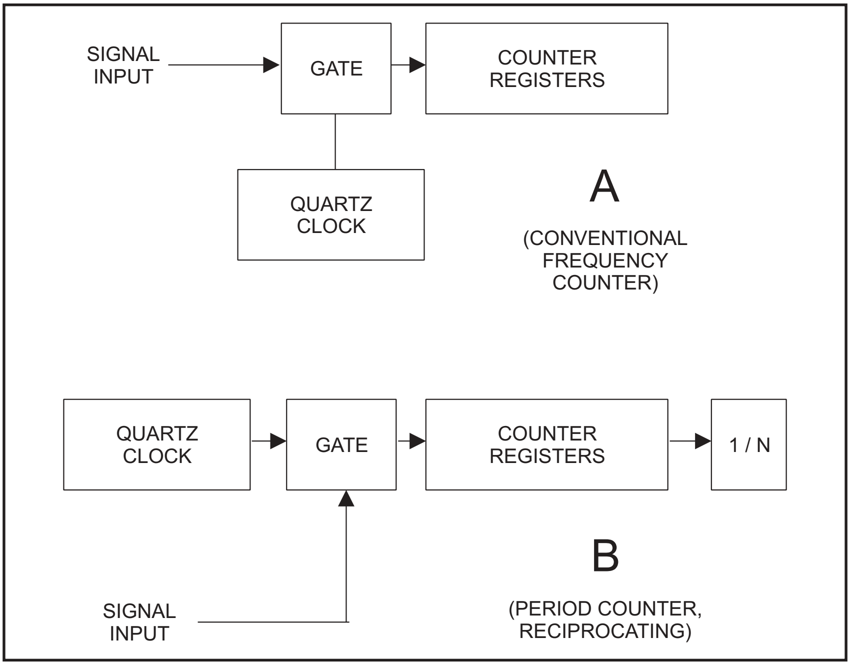

Figure 31. Block diagrams, conventional and period/reciprocal frequency counters.

the signal are accumulated into counter registers while the gate is open. If the gate is open precisely 1 s, the accumulated count is the signal frequency in cycles per second, or hertz (Hz). Other values of precisely-known gate time may be used if the counter register is appropriately scaled before display. Longer gate times are required in order to obtain better resolution.

In B, a "squared-up" version of the signal is used to control the gate and pulses from a quartz-based clock are accumulated into the counter. This count is proportional to the signal period; taking the reciprocal of the count provides a value proportional to the signal frequency, and scaling for the clock frequency provides the exact signal frequency. Whenever the signal frequency is lower than the clock frequency, this "period and reciprocate" technique of B always provides better resolution for any given measurement interval than the conventional counter architecture A. Since typical quartz clock frequencies are 10 MHz or higher, this is the preferred technique when measuring audio frequencies. It is also practical to hold the gate open for several or many cycles of the signal waveform in order to accumulate still more clock cycles into the register, if the register count is then divided by the number of cycles of waveform for which the clock was open. This period-averaging reciprocal counting technique is the most valuable for audio. It can permit selection of a desired measurement interval and then will provide maximum possible resolution and trigger noise rejection for the particular clock frequency and measurement interval in use.

Frequency Counter Instrument Criteria

Sensitivity of the input and triggering circuitry should permit frequency measurements on relatively low-amplitude signals, preferably down into the tens of millivolts or even lower. The reciprocal, period-averaging technique is by far the most useful at audio. Frequency accuracy will be proportional to the accuracy and stability of the quartz crystal oscillator, but actual audio applications can normally be satisfied with quartz oscillators far below the state of the art. Quartz oscillators with specifications within a few parts per million are readily available, but accuracies greater than a few hundred parts per million are rarely required for most audio work.

The basic frequency counter architecture has no frequency selectivity and will normally count the highest amplitude signal present or give false readings if the signal is complex. For certain audio applications, it is desirable to be able to pass the signal through a bandpass filter before connecting it to the frequency counter, allowing independent measurement of the frequencies of several components of moderately-complex signals such as IMD test signals.

Phase Measurement

Measurement of the phase difference between two audio signals of the same frequency is commonly required. Adjustment of the azimuth angle of stereo or multi-track analog tape recorder heads is best done by measuring the phase difference between the outside tracks and adjusting azimuth for zero phase. Stereo transmission paths must have matching phase characteristics if they are to properly preserve the intended stereo effect. Input-to-output phase shift of an amplifier is an important characteristic when designing feedback circuits. Phase is deliberately manipulated in several types of audio processing equipment.

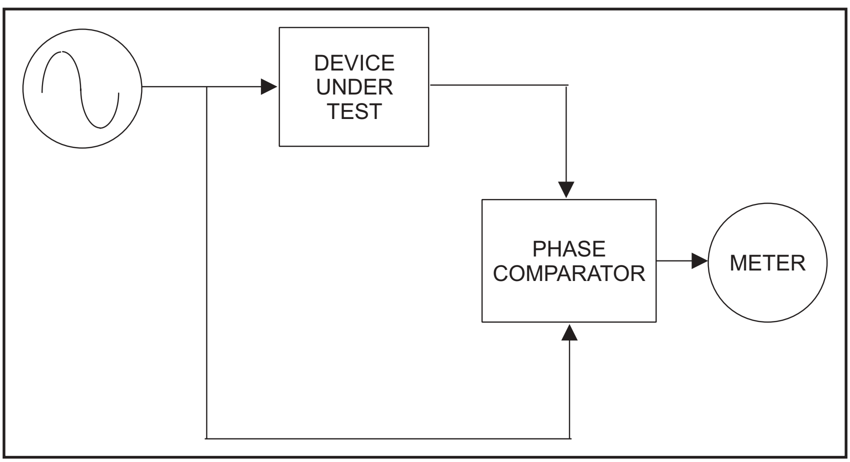

Figure 32. Input/output phase measurement.