communication link, the signal-to-noise (*S/N*) ratio at the receiver side suffers because the shifted audio signal contains more high-frequency components than the original (source-) signal. By contrast, when the first encoding system (Fig. 1a) is applied, the bulk of the intelligence is carried by components in the lower part of the spectrum, so that *S/N* degrading will not occur (in fact, there is psychoacoustic evidence that most information in speech and music is contained in the frequency range from 300 Hz to 1,000 Hz).

The system whose spectrum is shown in Fig. 1a yields almost unintelligible output signals, and is not easy to decode. The decoding process required to restore the signal to its original spectrum can be subdivided into a number of operations as illustrated in Fig. 2. As shown in Fig. 2a, the first function of the decoder is to limit the bandwidth of the encoded signal, i.e., to fit this into a frequency range from 500 Hz to 10 kHz. The filter slope at the high end of the spectrum prevents intermodulation products being generated in the decoding process, while the high-pass filter prevents whistles between 1 kHz and 2 kHz as a result of low-frequency components that are possibly added to the channel.

The block diagram of the decoder (Fig. 3) shows that the input signal is taken through a low-pass filter before it is amplified and applied to a high-pass filter. A switch connected ahead of the output buffer allows the user to select between encoded and non-encoded signals. When the switch is set to 'non-encoded', the input signal is fed direct to the output buffer, i.e., it is not amplified. This is done to ensure that the signal level of non-encoded signals is not changed by the decoder. Thus, the amplifier merely serves to compensate losses introduced by the phase filters used for encoding or decoding the input signal.

## Frequency transforms: Hilbert & Fourier

Shifting an audio spectrum can be achieved in three ways, which are familiar from radio communication techniques used for generating and demodulating SSB (single-sideband) signals. The first two systems are based on narrow-band filters. These are widely applied, and do not present problems with audio signals whose frequency range, for radio communication purposes, extends from 300 Hz to 3,000 Hz only.

Transmitting music via a communication link based on spectrum shifting poses more problems than speech because a much larger frequency spectrum must be conveyed without distortion. It is for this, and other, reasons, that the present encoding system is based on the so-called 'third method', a term familiar to most licensed radio amateurs and radio engineers (Ref. 3).

So how does the encoder work? First, the audio signal is split into two components. Both have the same amplitude, and contain the source signal. The only difference between these components is the phase shift between the frequencies they consist of. This phase shift is 90°, and one of the components is called the Hilbert transform of the other (the two signals are called a Hilbert pair). A Hilbert transform is a frequency shifting operation that unfortunately exists in theory only. In practice, however, a number of systems have been developed that yield close approximations of such an ideal transform. One of these systems is applied in the present encoder.

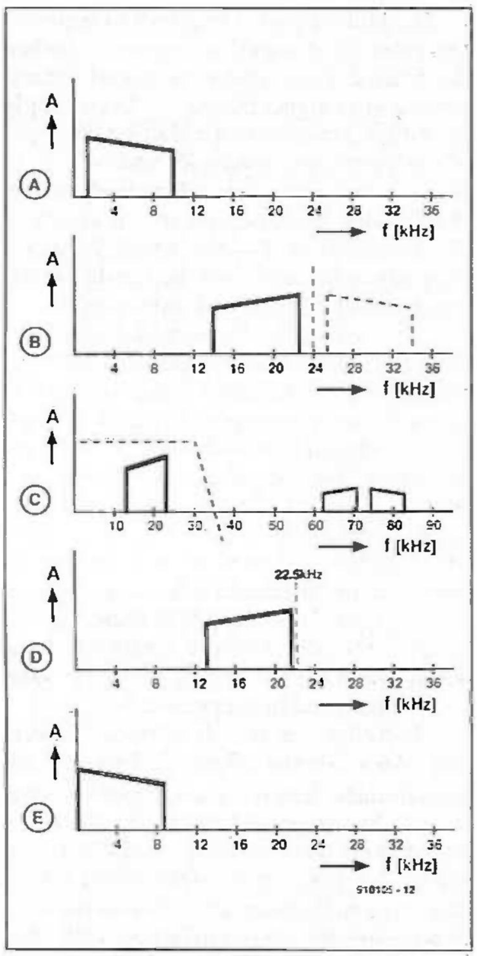


Fig. 2. Multiplication with two phase-locked carriers, filtering and sideband suppression are the main functions into which the decoding process can be subdivided.

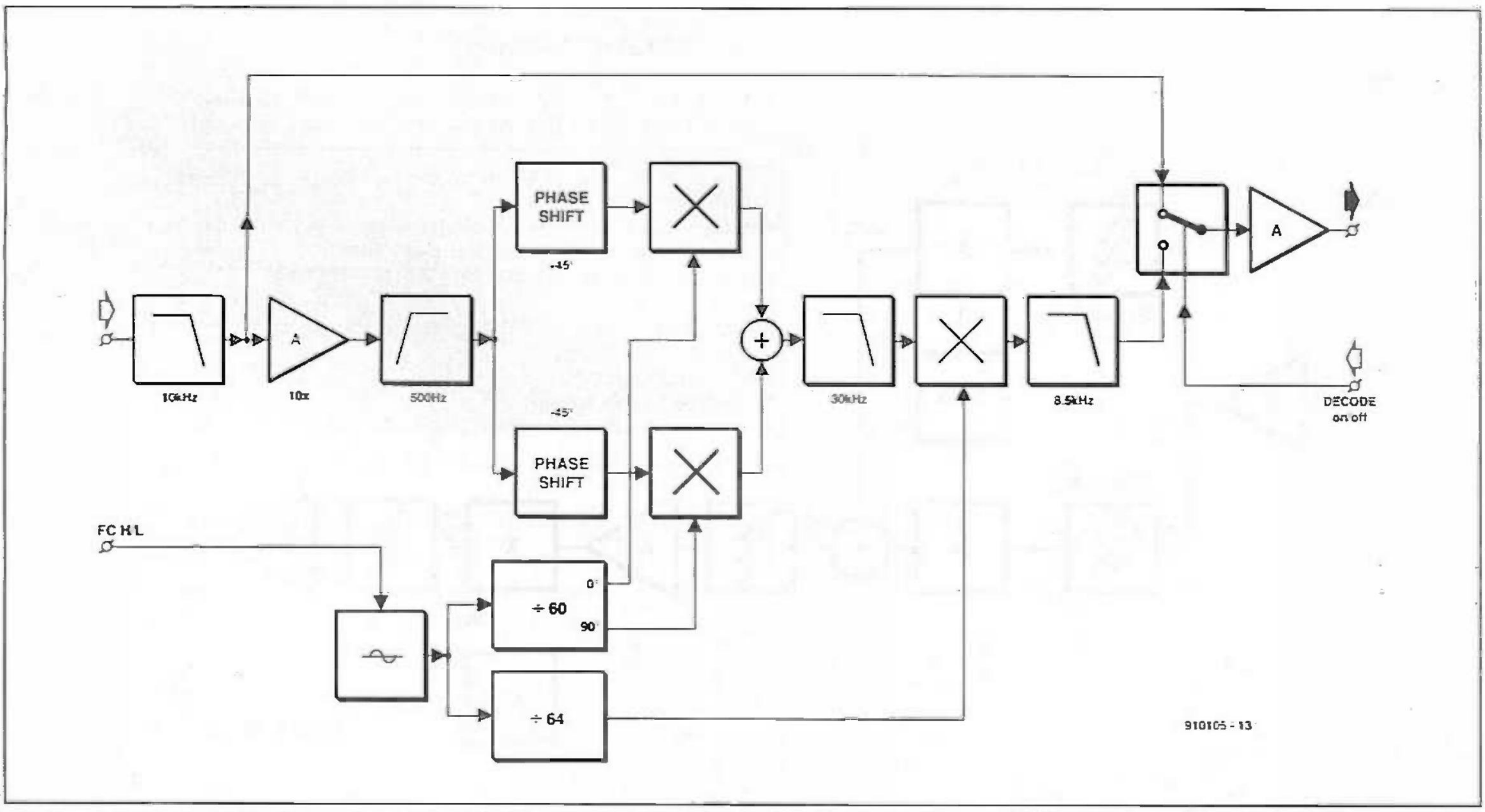


Fig. 3. Block diagram of the frequency shift encoder/decoder. The FC H/L input selects one of two system clocks from which the amount of frequency shift is derived. Note that the encoder/decoder has a bypass function, which is controlled via the S. 0/1 input.