

DSP Implementation of 300- and 1200-Baud FSK Packet Demodulators

Jon Bloom, KE3Z

American Radio Relay League
 225 Main Street
 Newington, CT 06111
 email: jbloom@arrl.org

ABSTRACT

This paper gives the theory of operation for a DSP implementation of **300-** and **1200-baud** FSK modems. Specific algorithms are described and filter characteristics are given for an actual implementation.

The use of digital signal processing (DSP) for packet modem implementation is clearly the technique of choice for speeds up to at least 9600 bauds. The declining cost of DSP chips makes DSP modem subsystems cost competitive with analog designs, and the increasing capabilities of these devices make design of packet modems straightforward at moderate speeds.

Packet modem standards have increased over the years, but the two earliest standards, **300-** and **1200-**baud FSK, have survived and continue in widespread use. The **1200-baud** standard, which is widely used with voice-grade VHF-FM radios, calls for tones at 1200 and 2200 Hz. (Note that packet sent using these standards is **NRZ-I** encoded, so the specification of one frequency as mark and the other as space is not necessary.) **300-baud** packet, used primarily with HF SSB radios, tends not to use a consistent pair of tones, since radio tuning can be used to adjust the received-signal center frequency as needed. A tone spacing of 200 Hz is used at 300

bauds. Implementing a phase-continuous FSK modulator for these standards is trivial, most easily done with a sine look-up table. Demodulation presents more substantial challenges, which this paper addresses.

Basic Demodulator

Demodulation is performed using the technique shown in Fig 1. The input signal sequence is delayed by a period equal to one quarter of the cycle (90°) of a signal at the center frequency of the signal being demodulated. For **1200-baud** FSK, this frequency is 1700 Hz, and this frequency is also used at 300 bauds for convenience. The delayed sequence is multiplied with the undelayed sequence. For a **sine-wave** input, the result is as follows:

$$A \sin(2\pi fnT) \cdot A \sin(2\pi fnT + d) = \frac{A^2}{2} [\cos(2\pi fd) - \cos(2\pi fd) \cos(2 \cdot 2\pi fnT) + \sin(2\pi fd) \sin(2 \cdot 2\pi fnT)]$$

(Eq 1)

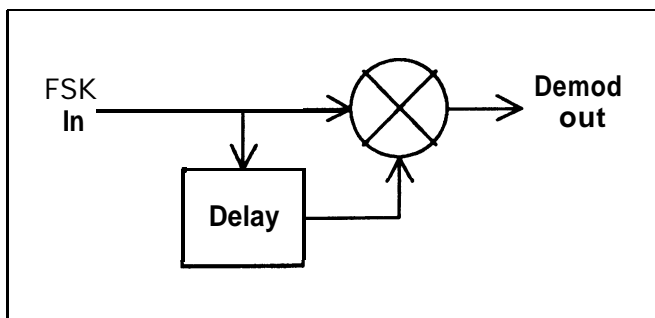


Fig 1-FSK Demodulator

where:

- A** is the peak input amplitude
- f** is the input signal frequency
- d** is the delay time
- T** is the sample period
- n** is the sample index

Note that this results in a constant term and two terms at twice the input frequency. These latter terms are filtered out, leaving the constant term:

$$\frac{A^2}{2} \cos(2\pi fd). \text{ For the case when } d \text{ is equal to } 90^\circ$$

at the input frequency ($d = 1/4f$), the constant term becomes zero. At frequencies higher than that, the result is a negative level, while at lower frequencies the result is positive. Thus the multiplication of the delayed and nondelayed sequence results in a constant term that describes the frequency shift of the input signal.

A Practical Demodulator

Examining the above equation shows that a fixed delay of one $1/4f_c$ is required, where f_c is the signal center frequency. The easiest way to provide this delay is to select a sample frequency at which an integral number of sample periods equals the required delay. The first such frequency for the **1700-Hz** center frequency of the modems described here is 6800 Hz. At that sample frequency, a **one-sample** delay corresponds to a delay of 90° at 1700-Hz. 6800 Hz is the sample rate used in this modem. Note that this allows input signals up to **3400-Hz** satisfying the Nyquist criterion. Systems in which

the designer is constrained to use a sampling rate other than one that provides this simple **integral-multiple** approach will require a filter that provides a suitable fixed delay. Such a filter is trivially implemented as a one-tap FIR filter.[1]

Fig 2 gives a block diagram of the complete demodulator. Since the output level of the demodulator described above depends on the input amplitude, an AGC stage **first** normalizes the input amplitude. An AGC level is computed and used to calculate a gain factor by which the input signal is multiplied. The input signal is first peak detected. That is, the amplitude of each input sample is compared to the current peak value. If the input value is greater, the current peak value is made equal to the input. If not, the peak value is multiplied by a value slightly less than 1 to decay it. This is shown in the flow chart of Fig 3. In the flow chart, $x(n)$ is the input sample, $y(n)$ is the output sample, and K is a constant chosen to give the desired output amplitude.

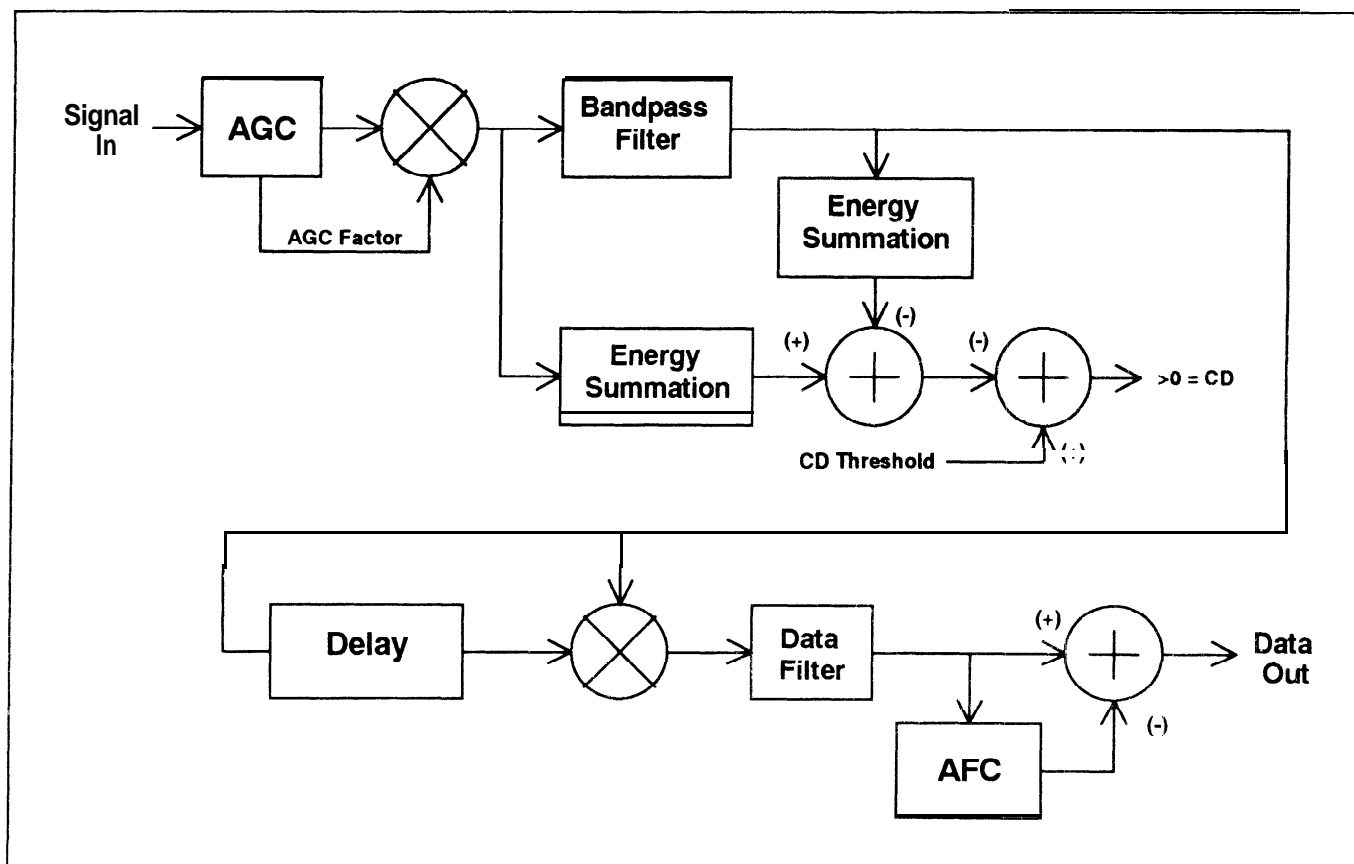


Fig 2-Demodulator Signal Flow Diagram

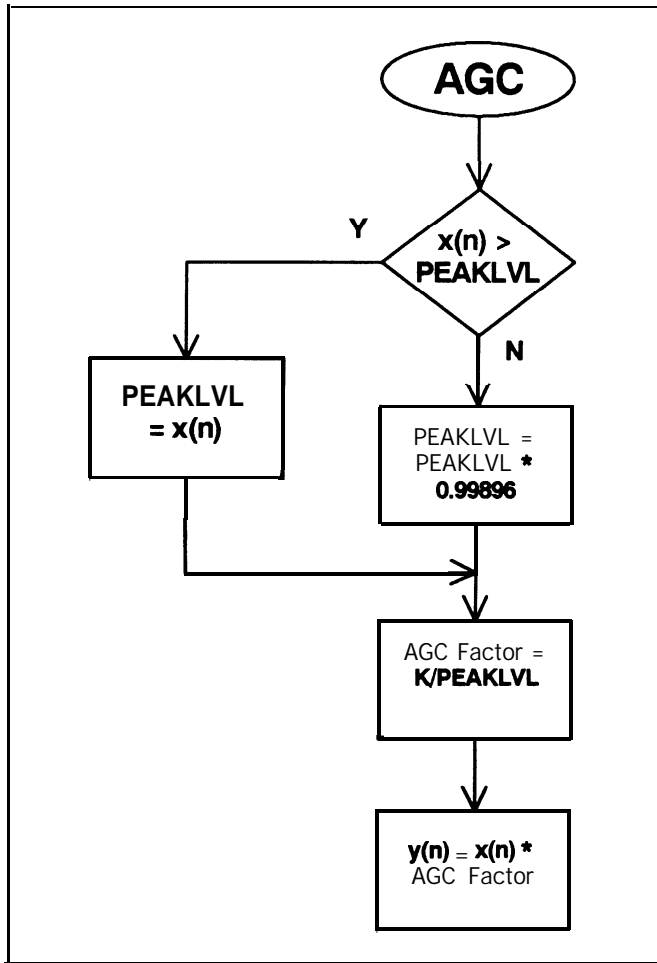


Fig 3-Flow chart of the AGC computation.

Carrier detection (CD) is done by comparing the energy in the input signal to the energy in a bandpass-filtered copy of the signal. When the input signal is composed of noise, much of the energy of the

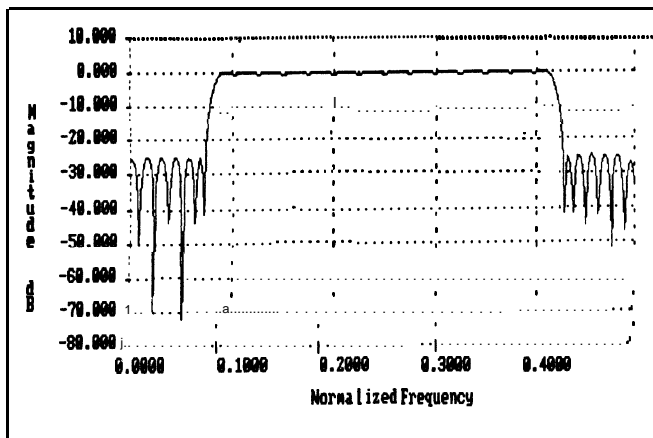


Fig 4-1 200-baud Bandpass Filter Response

signal will fall outside the **passband** of the filter. Thus there will be a substantial difference in the amount of energy in the two signals. But when the input signal is the desired FSK signal, the filtered and unfiltered signal will be nearly the same. Calculating the difference between the two signal energies and comparing the result to a fixed threshold develops a carrier-detect decision. The amplitude response of the **1200-baud bandpass** filter is shown in Fig 4, the **300-baud bandpass** filter in Fig 5.

Calculation of the energy of the signals is performed over blocks of 512 samples:

$$E = \frac{1}{512} \sum_{i=0}^{511} [x(i)]^2$$

At 6800 samples per second, this results in an updated CD result about every 75 milliseconds. Using 512 samples minimizes errors caused by having an aperiodic signal represented by the samples.

The bandpass-limited signal is applied to a detector like that described above. Band-limiting the signal ensures that no (unwanted) low-frequency signal present in the input audio produces a **sin(2·2πfnT)** signal (see Eq 1) that is in the **passband** of the data filter.

The detected signal is filtered by a low-pass filter that rejects signals above 600 Hz (**1200-baud** demodulator) or 150 Hz (**300-baud** demodulator). This filter removes the unwanted signal components from the detection process and removes any **high-frequency** noise on the detected signal.

An AFC process is used on the detected signal. If the FSK signal center is 1700 Hz, the positive and negative peaks of the detected waveform will be equal in amplitude (but negative in sign). If the signal center is not 1700 Hz, these peaks will be of different amplitudes. Note that this is **not** the same as saying there is a DC offset in the waveform. There may **be**—almost certainly there **will** be—a DC offset imposed by the data stream. Thus the AFC has to operate on the peak values of the detected signal rather than the average value. Attack and decay time constants are used to minimize AFC jitter. The AFC process is shown in the flickered of Fig 6. Note that it depends on there being continual data transitions, as are ensured by the use of HDLC packet framing. This technique would not be useful for an FSK system that did not ensure data transitions (e.g., asynchronous **RTTY**).

The AFC function can correct small tuning errors, although large offsets will cause part of the

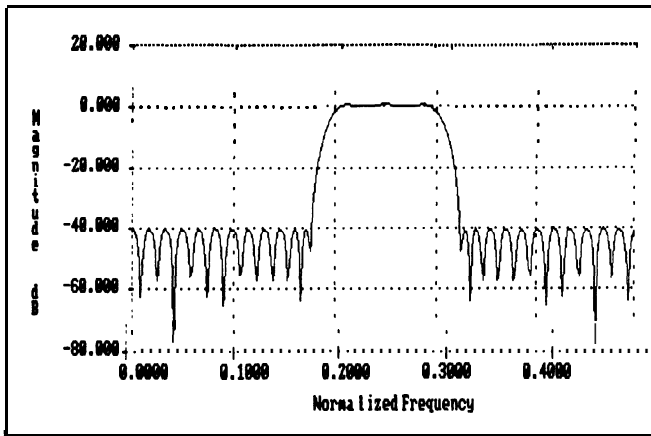


Fig 5—300-baud Bandpass Filter Response

signal spectrum to fall outside of the **bandpass** filter. AFC also helps during HF selective fading, as the reduction in amplitude of the fading tone causes the AFC to skew the decision point on the detected waveform. In the worst case, the complete absence of one of the tones, the detector becomes an AM detector, with the decision point being at one half the peak detector output.

Conclusion

While the modem described here has not been subjected to detailed testing, on-the-air experience and side-by-side comparison with several other packet modems has shown it to perform **well**. The author acknowledges that **300-baud** HF packet **and 1200-baud** VHF packet are not the best possible schemes. Still, DSP-based packet modems will likely need to include these modes to maintain compatibility with older, nonprogrammable packet modems. It is the author's hope that those who are implementing new DSP packet modem systems will be able to use the information contained in this **paper** to implement compatible modems with acceptable performance quickly and easily, the better to concentrate on development of newer, better modulation schemes.

Notes

[1] Phil Evans, "Implementation of an FSK Modem Using the TMS320C17," **Digital Signal Processing Applications with the TMS320 Family**, ed. Panos Papamichalis (Englewood Cliffs: Prentice Hall, 1990) **363-365**.

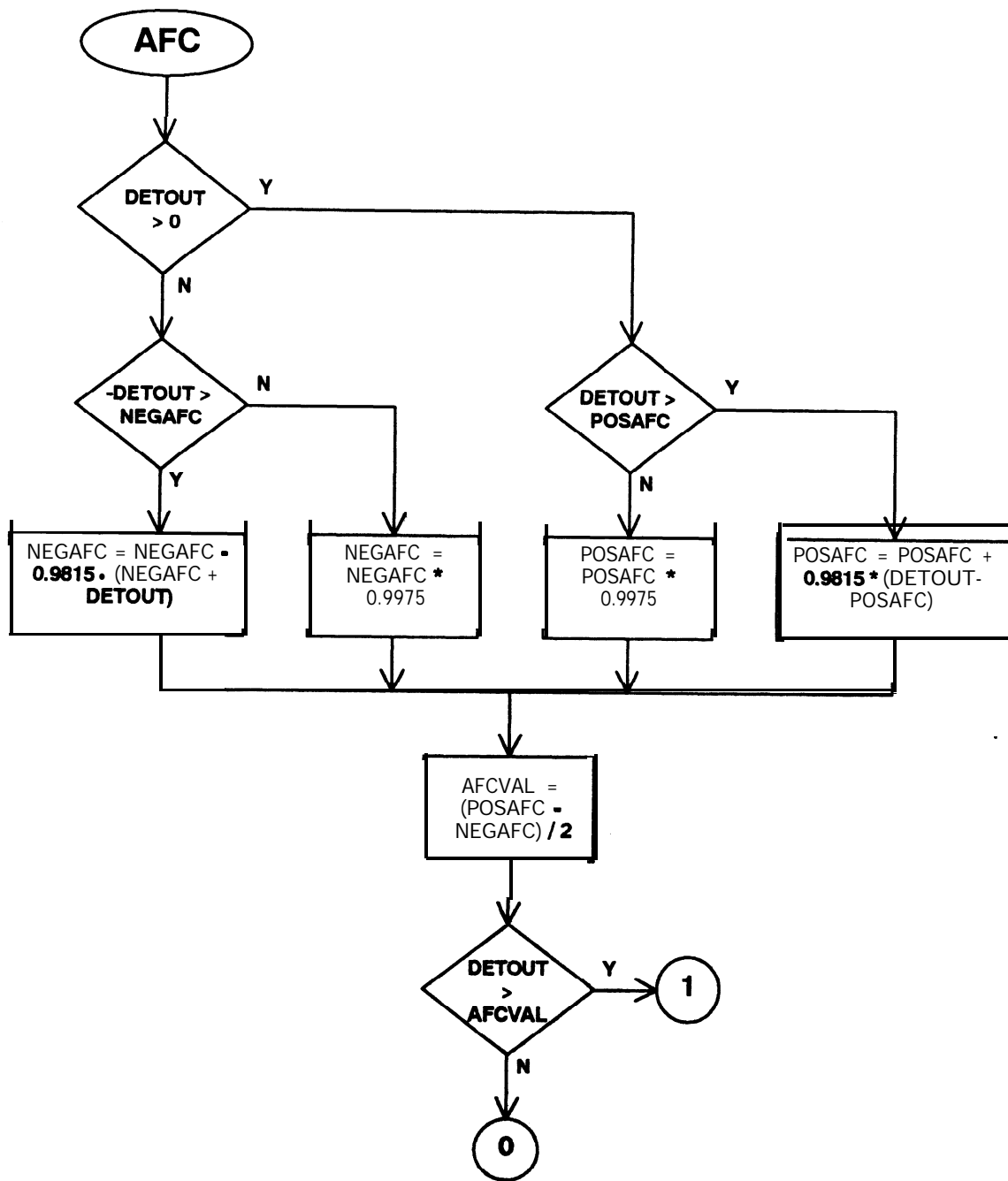


Fig 6—AFC Function Flow Chart