Frequency Enhanced Digital AM Radio
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Abstract - An all digital AM radio with stereo decoding is discussed in this paper. Digital demodulation, filtering and decoding are implemented as well as digitally compensating for the less than ideal frequency response of the typical AM analog receiver front end. Although the resulting stereo signal is not as good as that produced by FM stereo radio, it is a very good quality stereo signal and a noticeable improvement over standard analog reception techniques.

I. INTRODUCTION

Digital audio systems are popular in home audio systems and commonly found in today's more expensive automobiles. These systems use digital signal processing techniques to provide CD playback, concert hall effects, and noise cancellation. The same digital signal processor can also be utilized to perform the receive function of both AM and FM radio to produce a higher quality sound. Additionally, this would reduce the existing number of hardware components and cost which would make digital audio systems more affordable.

The authors derived and implemented an algorithm to perform AM radio reception with and without the C-QUAM® (Compatible Quadrature Amplitude Modulation) stereo signal. This paper outlines this derivation, analysis and implementation of the digital radio using digital signal processing techniques.

II. BACKGROUND

Although AM transmission for radio had been around long before FM and had a much larger share of the radio audience, the FCC approved a stereo standard for FM radio transmission in 1960 and has yet to specify one for AM radio transmission [1]. This has caused AM radio stations to lose much of their audience and popularity. Many techniques were developed to perform AM stereo transmission to recover a larger share of the radio audience.

The FCC did not specify one particular AM stereo technique as a standard as they did with FM, but did specify that every AM stereo broadcast technique must allow installed monaural AM receivers to receive an undistorted signal when tuned to a stereo channel. As much as this complicated the design of AM stereo techniques, the FCC tested and approved several standards including the C-QUAM technique in 1981[2]. The marketplace has made C-QUAM the defacto standard for AM stereo transmission with approximately 574 stations throughout the United States and 206 stations internationally for a total of 780 C-QUAM stations worldwide using C-QUAM encoding [3].

Due to the FCC imposed compatibility requirements, using a typical QAM system for transmitting L+R and L-R as two different signals as shown in Figure 1, is not possible. An envelope detector would not be able to detect this signal if a large modulation index existed because a stereo signal in which L only or R only was transmitted. Thus, it would not be compatible with existing receivers which use envelope detectors.

![Figure 1 Quadrature AM Stereo Transmitter](image-url)

To create a stereo signal, the C-QUAM technique transmits L (left channel) and R (right channel) information by encoding L+R and L-R channel information into the phase of the transmitter carrier. This "modified" carrier is then amplitude modulated by the L+R signal [4], as would be done in a typical monaural AM transmitter with an unmodified carrier. Figure 2 shows the carrier phase modification of a C-QUAM transmitter in block diagram form. The output of the I and Q modulators, respectively are:

\[ L+R \]
\[ L-R \]

\[ C-QUAM® is the registered trade mark of Motorola, Inc. \]
\[ I = [L(t) + R(t)] \cos(\omega_c t) \] (1)

and

\[ Q = [L(t) - R(t)] \sin(\omega_c t) \] (2)

Summing these signals results in

\[ \cos(\omega_c t) + [L(t) + R(t)] \cos(\omega_c t) + [L(t) - R(t)] \sin(\omega_c t) \] (3)

which can be expressed as

\[ \sqrt{[1 + L(t) + R(t)]^2 + [L(t) - R(t)]^2} \cos(\omega_c t + \phi(t)) \] (4)

where

\[ \phi(t) = \tan^{-1} \left( \frac{L(t) - R(t)}{1 + L(t) + R(t)} \right) \] (5)

and the output of the limiter is

\[ A \cos(\omega_c t + \phi(t)) \] (6)

where \( A \) denotes the pre-defined constant. This signal essentially becomes the carrier which is modulated by the \( L(t) + R(t) \) signal. The resulting signal from the transmitter is:

\[ T_x(t) = A[1 + L(t) + R(t)] \cos(\omega_c t + \phi(t)). \] (7)

It is possible to receive this on a non-stereo system with an envelope detector with little or no distortion. An envelope detector ignores the phase information contained in the signal [5], so that if the phase component is removed for the signal given above, the result will be

\[ A[1 + L(t) + R(t)] \cos(\omega_c t) \] (8)

which is the standard signal which would be seen by the receiver if the transmitter was monaural.

III. IMPLEMENTATION

Figure 4 shows a detailed diagram of the DSP based system. The input to the system comes from the AM receiver front end and tuning circuitry and has been downconverted to the 450 KHz IF frequency. The sampling rate at the IF frequency would have to be at least 900 KHz to meet the Nyquist criteria [6]. The AM signal is confined to a 10 KHz channel, however, so if the signal is further downconverted, a smaller sampling rate could be used to reconstruct the signal. As shown in Figure 5, the IF is
downconverted to approximately 25 KHz and a sampling rate of 4 times that is employed in the sigma delta A/D converter. The analog multiplier shown in Figure 5 creates all ranges of harmonic frequency contents above the fundamental carrier frequency. However, Sigma-Delta A/D converter is based on oversampling and decimation processing technology, it eliminates the need to use an anti-aliasing filter [7].

The decimator filtering consists of cascading two half-band filters, one with 11 taps and the other with 31 taps, each performing a 2:1 decimation stage. This is efficient in reducing the number of operations performed per input since the signal is downsampled to approximately 50 KHz in the first filter using only 11 taps. The second filter, which takes 31 taps, is performed at the lower sampling rate.

For decimation, this cascaded half-band filter structure has the following advantages: significantly reduced number of computations; reduced memory requirement; simplified filter design problem; and reduced finite-word-length effects [7].

Consider a cascaded (multi-stage) implementation of half-band filters. Figure 6 shows the block diagram of a 2 stage half-band filter structure. $F_1$ in Figure 9 denotes the sampling rate of the input signal to the decimation filter. Note that the decimation of 4:1 can be achieved by two 2:1 decimators in cascade.

Let's examine the lowpass filter frequency regions for the individual stages. For the $k$th stage decimator, when $k = 1, 2$, the lowpass filter's passband, transition band and stopband regions are
\[0 \leq f \leq F_k : \text{kHz stage passband} \]
\[F_p \leq f \leq F_k - F_p : \text{kHz stage transition band} \]
\[F_k - F_p \leq f \leq F_k : \text{kHz stage stopband} \]

(13)

where \(F_p\) and \(F_k\) are the passband frequency and the sampling frequency of the \(kHz\) stage output, respectively. Signal energy in the transition band will alias back upon itself (after decimation by 2:1) only from \(F_p\) up to \(F_k\). Hence the baseband, \(0 \leq f \leq 5f\), is protected against aliasing.

\[I = C\left[1 + L(k) + R(k)\right] \cos \gamma \quad (14)\]
\[Q = C\left[1 + L(k) + R(k)\right] \sin \gamma \quad (15)\]

and

\[\gamma = \tan^{-1}\left[\frac{L(k) - R(k)}{1 + L(k) + R(k)}\right] \quad (16)\]

Following the decimation filters are the filters which compensate the signal for the poor frequency response of the typical AM receiver front-end. A typical AM receiver front-end is analog and cuts the bandwidth to about 3.5 kHz as shown in Figure 7. This performance can be improved by applying a linear phase frequency compensation filter to the signal at the output of the 4:1 decimation filter stage. Since the frequency response of the compensation filter requires only monotone changes with respect to frequency, a simple (11-tap) symmetric FIR filter can compensate the frequency drooping to make a flat spectral response as shown in Figure 7. This effect can be enhanced by applying a low pass filter to detect the phase information resident in the \(Q\) signal. The result can then be filtered and used to generate the NCO input.
IV. VERIFICATION

The test system used to develop and verify the AM stereo is shown in Figure 8. The Leader AM Stereo Signal Generator is the heart of the test system and is designed to generate C-QUAM signals from baseband inputs. A CD player is used to generate the left and right signals used as input to the generator. The output of the generator is the feed to the input of an AM receiver front end chip which performs the tuning and downconversion of the signal to 450 Khz.

As described in section 3.0, the next stage does an additional downconversion and samples the signal for input into the DSP development system. The signal is then processed and the left and right channel information is output to an amplifier speaker combination. The development system is controlled by a host computer to allow for efficient development and testing of the design.

V. CONCLUSION

An expensive analog receiver available today typically produces 40 dB of channel separation. The digital implementation discussed in this paper achieves more than 40 dB of channel separation and is just the initial implementation. Additional enhancements which are anticipated to improve the signal will be implemented and verified. The DSP implementation provides a flexible and higher fidelity solution for AM radio reception.

REFERENCES