Capsim Application Note

Voiceband Codec with μ-law Companding

Introduction

This application note describes the modeling and simulation of a voiceband μ-law codec. In particular the action of the compander, the sampling process, and filtering will be investigated.

System Description

The top level block diagram of the system is shown in Fig. 1. The system consists of a sine wave generator, a transmitter, a receiver, a signal to total distortion measurement block, probes, and an x-y display. The galaxy for the transmitter is shown at the top. The receiver galaxy is shown below the top level system.

The sine wave generator has a number of parameters through which the amplitude, frequency, sampling rate and number of samples to generate may be modified. For a companding codec, we are interested in the signal to noise ratio as the input amplitude is decreased from the peak level down to -55 dB's.

The transmitter galaxy, *TxCodec*, first filters the input signal with a bandpass IIR elliptic filter. This filter eliminates 60 Hz interference and also bandlimits the signal to prevent aliasing. The passband ripple, stopband attenuation, transition width and other filter parameters can easily be changed during multiple simulation runs.

The filtered signal is sampled by a sample and hold block. The sampling rate for the system is set at 64 kHz. This is 8 times larger than 8 kHz which is the sampling rate of the codec. The sample and hold(S/H) is clocked by a pulse generator. The pulse generator produces a periodic pulse at 8 kHz(adjusted by specifying the period to be 64/8 samples). The S/H block samples its input signal on the rising edge of the clock. This block has a parameter which permits the hold time to be adjusted. In the simulations to follow, the hold time is half the clock period or 62.5 micro-seconds.

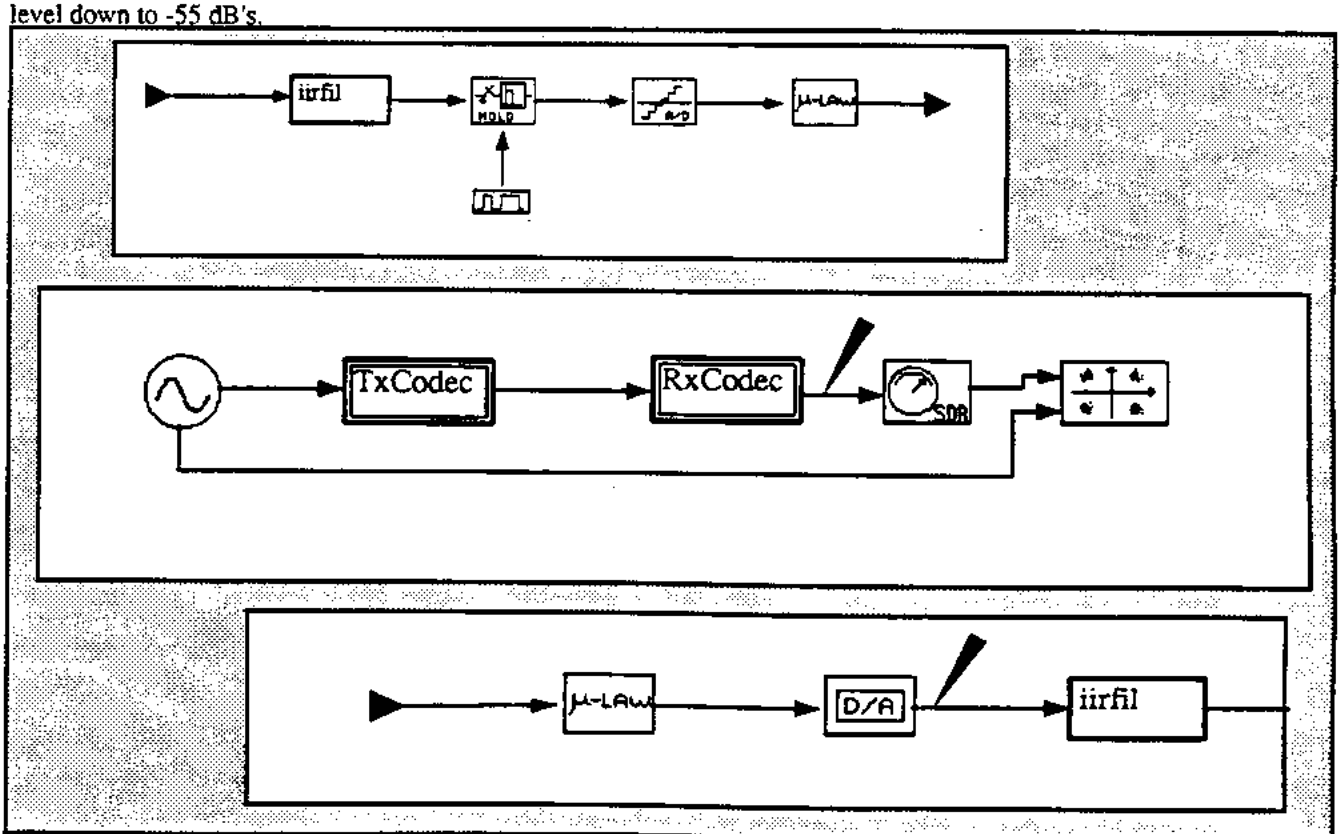


Figure 1.

The sampled signal is then converted to digital format through a uniform analog to digital converter. The parameters to this block include the maximum and the minimum input voltage and the number of bits used to quantize the input sample. For the voice-band codec, this is set to 13 bits. The A/D output is compressed using a μ-law compressor. Thus the 13 bit samples are compressed to 8 bits. The bit rate is now 64 kbits/sec (8* 8kHz).

The 8-bit samples are mapped to 13 bit uniform samples by the μ-law expander (actually the same star as in the transmitter but with the parameter changed to expand). The 13-bit samples (with μ-law quantization noise) are converted to analog by the digital to analog converter. The signal is then lowpass filtered to reproduce the transmitted waveform.

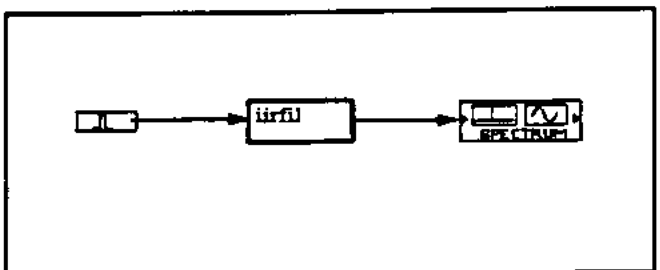


Figure 2. Filter Toplogy

Filter Characteristics

We can examine the filter characteristics of the receive lowpass filter (and transmit filter) by constructing the topology shown in Fig. 2. The *spectrum* star plots the time domain and frequency magnitude spectrum. Fig. 3 shows

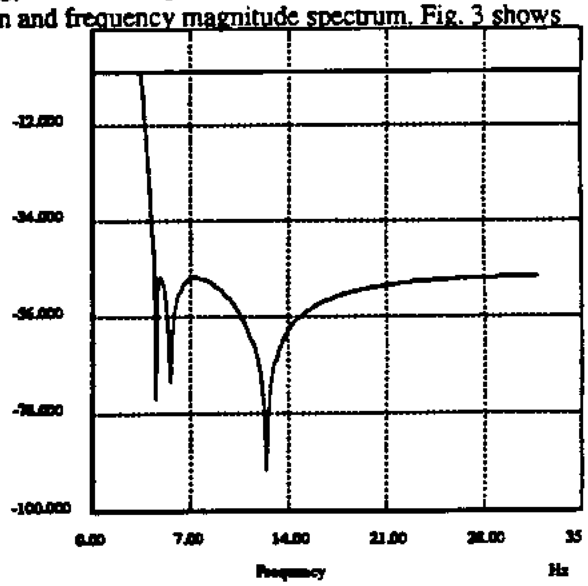


Figure 3. Receive Filter Frequencty response

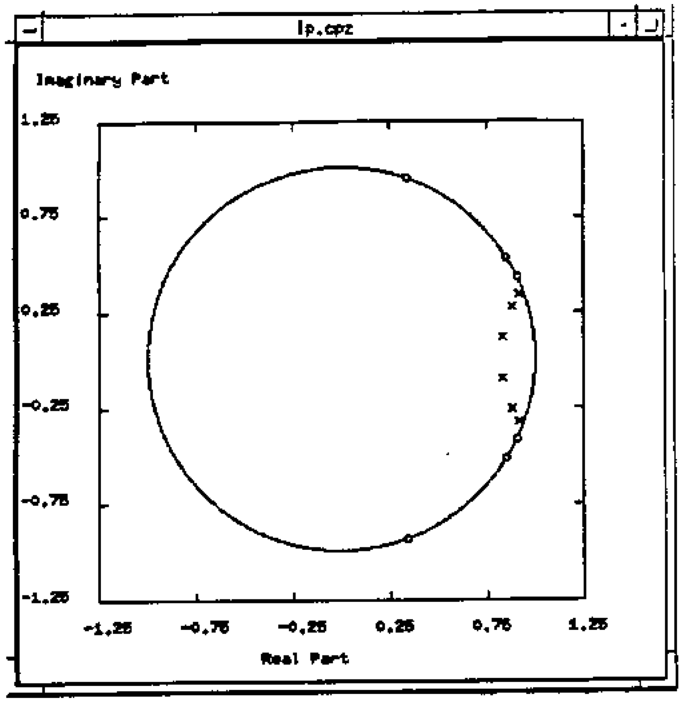


Figure 4. Receive filter pole-zero distribution

the magnitude spectrum of the low-pass elliptic receive filter. Note the cutoff frequency around 3.4 kHz. The pole-zero distribution of the low pass filter is shown in Fig. 4. To observe a close approximation to the impulse response of a voice channel, we show the impulse response of the transmit filter is shown in Fig. 5.

To see the receive filter in action, refer to Fig. 5 and Fig. 6 which show the time domain and frequency domain spectrum of the D/A output. The filtered signal is shown in Fig. 7. Note that the half width sampled signal in Fig. 5

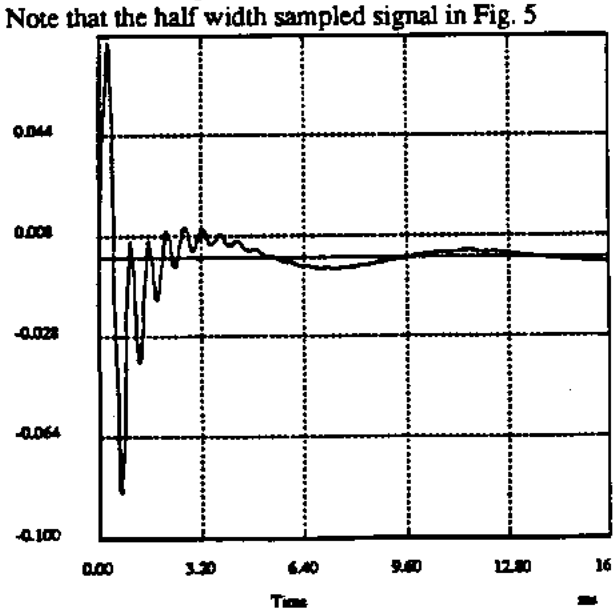


Figure 5. Transmitt Filter impulse response

has been smoothed by the filter. The tone is at a frequency of 1004 Hz.

The spectrum of the output of the digital to analog converter shown in Fig. 7 contains a wealth of information about the codec. First notice the tones at (7,9), (15, 17), (23,25) kHz. These tones are produced in the sampling process and occur centered on 8, 16, 24 kHz. The µ-law

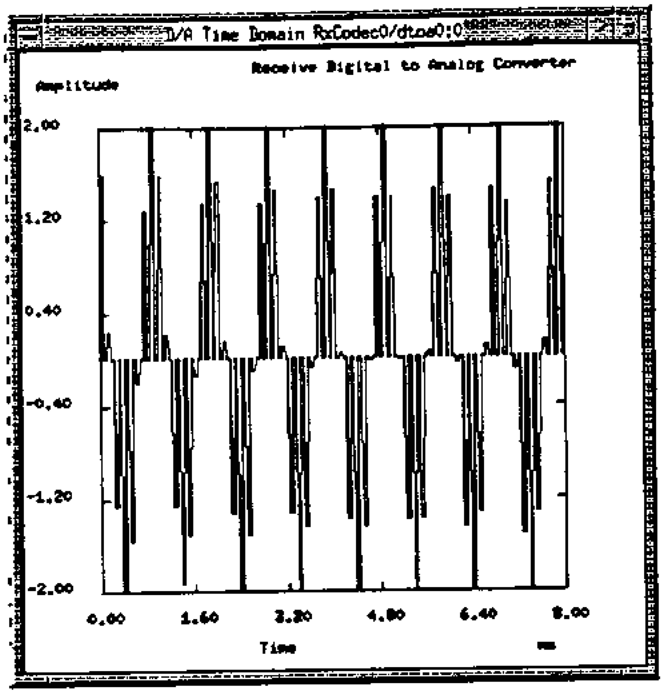


Figure 6. Analog waveform at receive digital to analog converter output

quantization noise is present as broadband noise. Also, the magintude of the tones follow a sin(x)/x envelope, as does the broad band noise. Notice the null at 16 kHz. This is the first zero crossing of the sin(x)/x response for the 62.5 microsecond hold time for each sample (see Fig. 6).

Companding

Next we investigate the companding action of the codec. The top level block diagram contains a star to measure the signal to total noise and harmonic distortion. As the input level is decreased from the 2 volt peak level, we measure the SDR. The results of these simulations are tabulated in the table below. The results clearly show that the signal to noise ratio remains constant over a wide dynamic range. The compressor at the transmitter uses finer quantization for small signals and coarser quantization at larger input levels. Thus, the noise decreases as the input amplitude decreases, maintaining a constant signal to noise ratio.

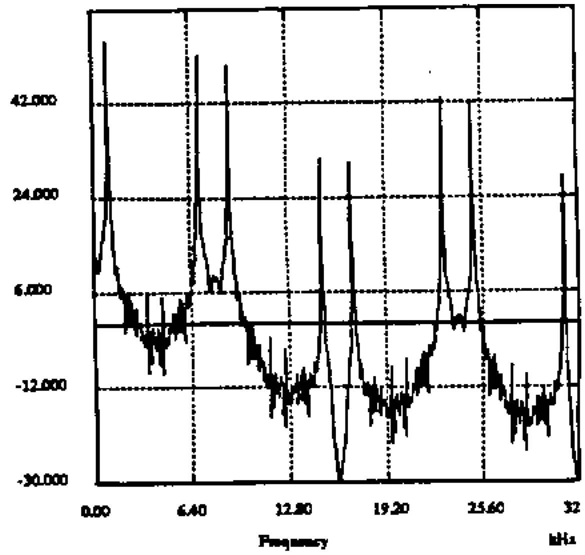
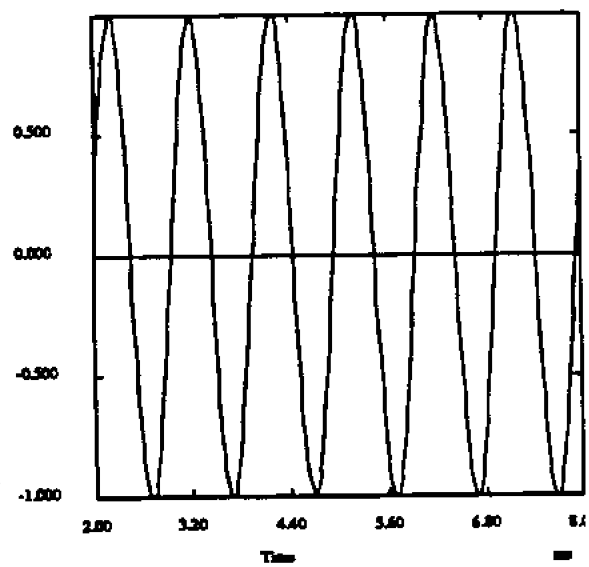
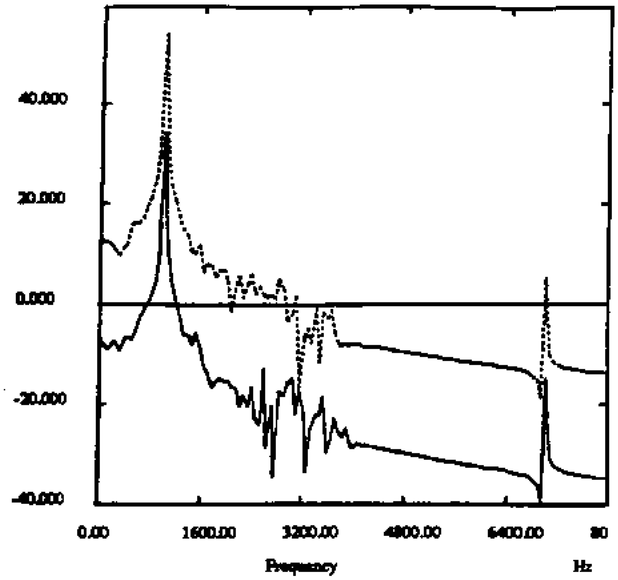


Figure 7. D/A output magnitude spectrum

Input level (dB) Relative to peak	SNR (dB)
0	39.27
-5	37.5
-10	38.0
-15	37.2
-20	38.35
-30	34.86
-40	31.86
-50	25.15



Fiugure 8. Filtered receiver output.



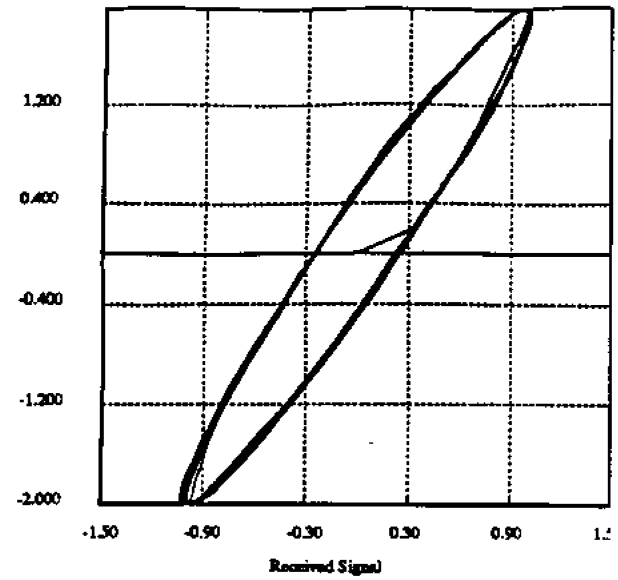


Figure 9. Comparison of received signal spectrum for peak level and -20 dB below peak

Figure 10. X-Y plot of input signal versus outppput signal for phase measurement.

The compander action is clearly illustrated in Fig. 9 which compares the spectrum of the received filtered signals for input amplitudes of 2 volts (the peak level) and 0.2 volts (-20 dB below peak). The noise decreased by 20 dB as the input level was decreased by 20 dB.

The phase response (and, therefore, the group delay) can be obtained by observing the x-y display of the transmitted and received sine wave. The top level block diagram in Fig. 1 shows the connections to the *scatter* star. The *scatter* star parameter was changed so that a continuous curve is obtained. The result at 1 kHz is shown in Fig. 10.

Conclusions

This report illustrates the usefullness of Capsim as a simulation tool for investigating the behaviour and for modeling a voiceband codec. The simulator easily models an inherently analog system through over-sampling. Hopefully, the ideas presented in this report will aid engineers in their efforts to simulate communication links.

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All topologies and eye diagrams in this report were produced by Capsim in PostScript.