An aliasing reduction method for hard-clipped sampled signals is proposed. Clipping in the digital domain causes a large amount of harmonic distortion, which is not bandlimited, so spectral components generated above the Nyquist limit are reflected to the baseband and mixed with the signal. A model for an ideal bandlimited ramp function is derived, which leads to a postprocessing method to reduce aliasing. A number of samples in the neighborhood of a clipping point in the waveform are modified to simulate the Gibbs phenomenon. This novel method requires estimation of the fractional delay of the clipping point between samples and the first derivative of the original signal at that point. Two polynomial approximations of the bandlimited ramp function are suggested for practical implementation. Validation tests using sinusoidal, triangular, and harmonic signals show that the proposed method achieves high accuracy in aliasing reduction. The proposed 2-point and 4-point polynomial correction methods can improve the signal-to-noise ratio by 12 and 20 dB in average, respectively, and are more computationally efficient and cause less latency than oversampling, which is the standard approach to aliasing reduction. An additional advantage of the polynomial correction methods over oversampling is that they do not introduce overshoot beyond the clipping level in the waveform. The proposed techniques are useful in audio and other fields of signal processing where digital signal values must be clipped but aliasing cannot be tolerated.