This is the algorithm that calculates SNR,THD,SINAD,ENOB,by using the following formulas:

**SNR:**

$SNR_{THEORETICAL}=6.02*N+1.76$ (dB) (1)

$$SNR = S - NOISE\ FLOOR - 10\log_{10}(M_T/2) \quad (2)$$

**THD:**

$$THD = 20\log\left(\sqrt{10^{[V_2/20]2} + 10^{[V_3/20]2} + .. + 10^{[V_{10}/20]2}}\right) = 10\log\left(10^{[V_2/20]2} + 10^{[V_3/20]2} + ... + 10^{V_{10}/20\,2}\right) \quad (3)$$

**SINAD:**

$$SINAD_{THEORETICAL} = \frac{P_{signal} + P_{noise} + P_{distortion}}{P_{noise} + P_{distortion}} \quad (4)$$

$$SINAD = 20\log\left(\sqrt{10^{[-SNR/20]^2} + 10^{[-THD/20]^2}}\right) \quad (5)$$

**ENOB:**

$$ENOB = \frac{SINAD - 1.76 + factor}{6.02} \quad (6)$$

where factor is:

$$factor = 20\log\left(\frac{full\,scale\ amplitude}{real\,amplitude}\right) \quad (7)$$

The algorithm takes inputs from files. The inputs should be digitized samples and the size of the file should always be a power of two.
I supposed that the sampling frequency of the ADC is 100MHz but if the sampling frequency changes,lines 36 and 92 should be modified.
A short explanation of how the algorithm works is the following;

The algorithm takes the Fourier Transform of the digitized samples,converts all the quantities into dB units and works in frequency domain. It uses real samples from the ADC .Firstly,it uses a vector in which it appends the values of the digitized samples. Also,a window function is used. It is multiplied it with the digitized sine wave and then the FFT of that multiplication is calculated. This is useful in order to make the important parts of the FFT clear so it would be easier to discriminate the signal,noise and distortion from each other.  Bartlett function is used for that purpose but Hamming,rectangular or even some other window function could be also used. It should be noted here that in order to obtain spectrally pure results,the FFT data window must contain an exact integral number of sine wave cycles,otherwise spectral leakage would occur.

To detect the harmonics and to calculate the distortion,two vectors are used;one to append the results of  the FFT of the multiplication of the window function and the digitized-sine wave and one to keep the corresponding frequencies. The condition used in order to recognize a value as a peak or not was double; firstly,the algorithm searches for local maximums of the FFT. Then,it recognizes them as distortion  only if the corresponding to those peaks frequency value is close to a value that equals to an integer multiplication of the sine wave' s frequency.”How”close depends on the number of samples. The distortion is then calculated by using eq.(3).

Following,the noise is now easy to measure;anything that isn't distortion is recognized as noise. Of course,the pure sine wave 's component is excluded. So,the mean value of those components is the noise floor. This is how SNR could be calculated by using equation (2).

The calculation of ENOB and SINAD was than easy to do by using equations (5),(6),(7).