Speech enhancement algorithms aim to reduce background noise of a corrupted speech input without distorting the original clean speech. In real-world situations, this can be very challenging. Although many algorithms have been developed to improve the Signal-to-Noise ratio (SNR) of the noisy input, they also introduce speech distortion and artefacts such as musical noise, damaging speech quality and intelligibility. Recently, there has been considerable evidence, from both physiological and speech perception standpoints, to support the use of the modulation domain for speech enhancement, where the modulation domain refers to the temporal variations of the acoustic spectral components. This report proposes modifications to existing modulation-domain speech processing methods, where an Ideal Binary Mask (IBM) will be applied on training samples of noisy speech to obtain averaged statistical information that can then be applied on new test samples. The goal is to use this information to improve the performance of an existing modulation-domain Kalman Filter (MDKF). The performance of these proposed modifications is assessed by measuring the SNR, speech quality (using Perceptual Evaluation of Speech Quality or PESQ) and intelligibility (Short-Time Objective Intelligibility or STOI) of the enhanced speech. Over a range of input noise levels, the developed algorithms have been found to give some improvements in both speech quality and intelligibility.