Speech enhancement algorithms aim to reduce background noise without distorting the original speech signal. 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 signal, they also produce 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 existing Ideal Binary Mask (IBM) or Target Binary Mask (TBM) will be applied on training noisy speech samples to obtain averaged statistical information that can then be applied on new test samples. The goal is to use this information to modify an existing modulation-domain Kalman Filter (MDKF) to improve on its performance. The performance of these proposed modified algorithms is assessed by measuring the SNR and speech quality (using PESQ) and intelligibility (using STOI) of the enhanced speech. By all measures, over a range of input noise levels, the developed algorithms have been found to give ?? improvements.