

FINAL YEAR REPORT

RF SPEECH PROCESSOR

project report presented in partial fulfillment
of the requirements for the award of Diploma in
Electrical Engineering (Electronics) of MARA
Institute of Technology.

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1.0 INTRODUCTION

1.1 General Concept

Spoken voice links are one of the least efficient ways of communicating information electrically, but the most natural to us humans. The classic compromise is to clip or to compress the speech signal into smaller bandwidths.

There are various ways of obtaining improved performance. The performance can be improved by the increasing the so-called 'talk power' of the signals. This really means making the signal as effective and powerful as possible within given peak amplitude limits. Most voice links have some form of processing to boost performance, even it is only in the form of some simple filtering. Bass frequency do not aid intelligibility to a significant degree and can even hinder it to a limited extent. Removing bass frequencies enables the remaining signal to be boosted slightly without giving any increase in the peak amplitude, and this makes it slightly more effective. Another benefit is that reduced bandwidth can be used at the receiving equipment, making it slightly less vulnerable to problems with noise and general interference.

Some high frequency components do significantly aid intelligibility, but using low pass filtering with a cut off frequency at about 3kHz or a little less does not greatly hinder the clarity of the signal, and the removal of these frequencies again enables the remaining signal to be boosted without the signal