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SYNCOMPEX – A VOICE PROCESSING SYSTEM FOR LOW COST HF RADIO TELEPHONY

by

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ABSTRACT

"Syncompex" is an acronym for "synchronized compressor and expander" and is a system designed to be used with HF single-sideband radio telephone equipment to improve its performance in the presence of interference and fading. It can be used either as an integrated part of a radio design or as a retrofit to existing equipments. The concept is functionally similar to Lincompex but is implemented with modern technology to reduce its cost and to allow its use in conjunction with commercial grade radios.

Preliminary evaluation of this system over two HF skywave paths is described. Significant improvements in voice quality and intelligibility were observed during periods of low signal-to-noise ratio.

1. INTRODUCTION

Radio signals in the High Frequency (HF) band propagate over long distances by reflecting from the ionosphere which is dynamic in nature. This causes signal fading, and, coupled with the existence of strong interference in the HF portion of the spectrum, often makes the received voice difficult to understand and uncomfortable to listen to.

The British Post Office developed the Lincompex system during the mid 60's (Ref. 1) to overcome the above mentioned problems. Although Lincompex has proven to be successful in international radio telephone circuits, HF users with limited financial means have not been able to take advantage of it because of its high cost and because it required the use of radios with an end-to-end frequency offset of less than ±2 Hz.

Syncompex was designed specifically with small users in mind. Digital techniques are used throughout to take advantage of the potential cost saving offered by recent advances in microprocessors/LSI technology. Syncompex is designed to operate with an end-to-end frequency offset up to ±20 Hz. The RF spectrum required is the same as for a normal SSB transmission. This allows Syncompex to be used with standard radios and also to be retrofitted to many older radios already in service.

Tests in which ionospheric fading and high ambient noise were encountered showed that a substantial improvement in signal-to-noise ratio can be attributed to the use of Lincompex (Ref. 2). As expected, a similar degree of improvement has been achieved with Syncompex. Experimental results obtained over HF paths of 40 miles and 700 miles, are described in Section 4.

2. PRINCIPLE OF OPERATION

The peak-to-average power ratio of a standard SSB radio telephone is about 14.5 dB. This figure is obtained when saturation of the power stage is allowed about 1% of the time (Ref. 3). This means that the maximum power capability of the transmitter is rarely attained when voice is being transmitted. When noise is encountered, the high-level syllables of the voice can be understood but the low-level syllables are masked, degrading the intelligibility of transmitted voice.

The Syncompex system circumvents the above mentioned difficulty by increasing the amplitude of the low-level syllables so that the dynamic range of the transmitter wave form is reduced, allowing all the syllables to be transmitted at or near full power. In this way the low-level syllables can be heard as well as the high-level, thus improving the voice quality at the expense of increased average power. (In the experimental Syncompex unit the increase in average power is measured to be in the order of 5-10 db).

The system consists of two variable gain audio amplifiers, one attached to the transmitter called the compressor, and one attached to the receiver called the expander. The function of the compressor is to reduce or compress the dynamic range of the voice. This is achieved by increasing the gain when a low-level syllable is encountered. The function of the expander is to perform the opposite function, that of increasing or expanding the dynamic range, thus restoring the voice to its original form. This process requires the expander and the compressor gain changes to be time synchronized, and the amount of expansion of the receiver to be the reciprocal of the amount of compression in the transmitter. (i.e. when the instantaneous gain of the compressor is N, the instantaneous gain of the expander is 1/N). The total system gain of the Syncompex is always unity when synchronized and is therefore transparent to voice and other analog signals passing through it. By allowing a radio system to operate in a mode approximately "constant net loss", Syncompex provides a feature important when connection with the switched telephone system is required.

The instantaneous gain of the compressor is limited to $N=2^Y$, $Y=0, 1,2...7$ and is held constant during an entire syllabic interval. Therefore it is necessary to transmit one value of Y per interval to maintain synchronization of the expander. This is achieved by transmitting a digital signal using a frequency modulated carrier in portions of the audio spectrum pre-empted from the normal voice signal. The bit timing of the digital signal is used to define the precise time at which the gain change in the compression process occurs. Because of its functional importance, in-band frequency diversity is employed to protect the digital signals from path disturbances. The digital signal is referred to as the control channel.

Syncompex improves voice quality because low-level syllables are reproduced in the receiver with the same fidelity as the high-level syllables. This effect is most noticeable during speech pauses in which the expander gain is reduced to a minimum, suppressing the noise during those intervals.

The digital nature of the control channel accounts for the relative insensitivity of the Syncompex system to a limited amount of frequency misalignment between the transmitter and the receiver. The system can operate normally as long as the frequency off-sets do not disable the digital modulation process. This limit was experimentally determined to be ±20 Hz in the existing hardware models. It is possible to use the control channel to obtain a precise alignment of the receiver with the transmit frequency if this is considered to be important. Since the location of the control channel in the spectrum relative to the carrier is precisely defined, misalignment of transmit and receive frequencies will cause a corresponding shift of the control channel. This principle has been used to implement a visual indicator to facilitate adjustments of the "clarifier" of the SSB receiver. The degree of frequency alignment that can be achieved using the indicator exceeds the control channel requirement and eliminates all traces of distortion in speech due to clarifier mis-adjustment. The process can be automated in a system in which all radios are equipped with Syncompex.

3. SYSTEM BLOCK DIAGRAM

A detailed diagram of the Syncompex is shown in Figures 1A and 1B.

3.1 TRANSMITTER

The audio input, after appropriate filtering, is sampled by an A-D converter operating at 9600 samples per second under the control of a microprocessor. The samples are stored in the microprocessor memory in blocks of 128. An algorithm incorporated into the microprocessor program memory operates to change the gain of each block of 128 samples (defined here as a syllable with duration 13.33 milliseconds). The gain is limited to $N=2^Y$ where $Y=0,1,....7$, and the change in gain is limited to 6 db per syllable. (Change in Y between adjacent syllables is ±1). Under the above mentioned two limitations the microprocessor applies the largest gain possible without causing the output D-A converter to saturate excessively. The D-A output is filtered forming the compressed voice output.

A: Microphone
B: Bandpass filter 300-2800 Hz
C: Analog to digital converter
D: Microprocessor
E: Digital to analog converter
F: Band elimination filter
- Stop bands at 765 and 2125 Hz
G: FSK modulator 765 ± 42.5 Hz
H: FSK modulator 2125 ± 42.5 Hz
I: Bandpass filter 765 ± 125 Hz
J: Bandpass filter 2125 ± 125 Hz
K: Hybrid
L: Hybrid

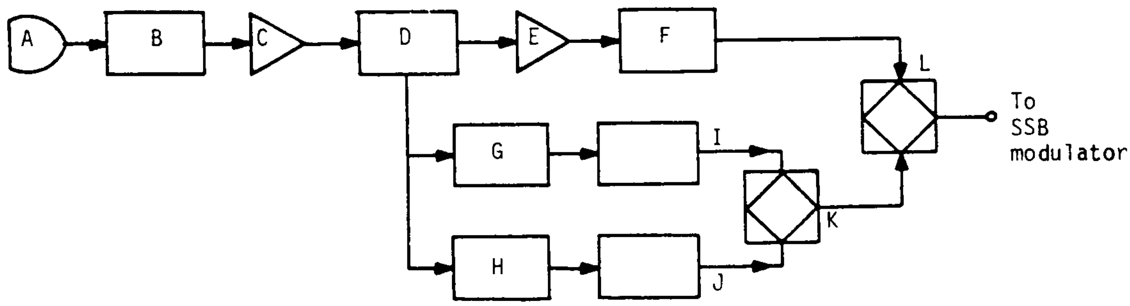
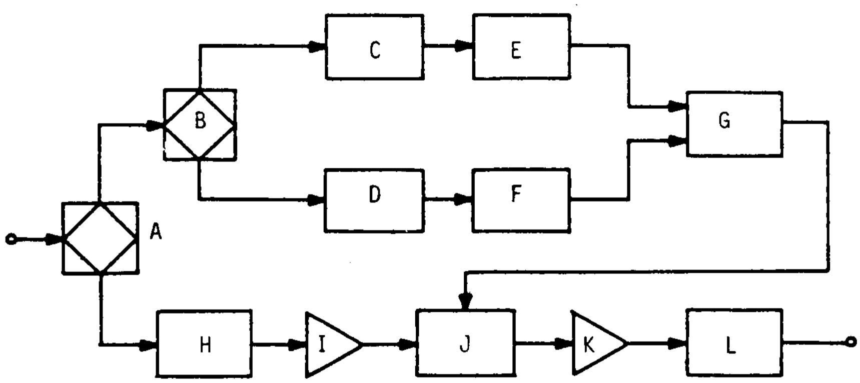


Figure 1A. Block Diagram of Syncompex Compressor



A: Hybrid
B: Hybrid
C: Bandpass filter 765 ± 125 Hz
D: Bandpass filter 2125 ± 125 Hz
E: FSK demodulator 765 ± 42.5 Hz
F: FSK demodulator 2125 ± 42.5 Hz
G: Diversity Combiner
H: Band elimination filter
- Stop bands at 765 and 2125 Hz
I: Analog to digital converter
J: Microprocessor
K: Digital to analog converter
L: Bandpass filter 300-2800 Hz

Figure 1B. Block Diagram of Syncompex Expander

The gain change applied by the microprocessor is used as an input to a pair of FSK modulators operating at 75 bits per second. The FSK modulators use ±42.5 Hz shift between mark and space and are centered at 765 and 2125 Hz. Since the gain change is limited to a 6 dB step every 13.3 milliseconds, a binary channel of 75 bits per second is sufficient to carry the information required and allow the expander to track the compressor.

Identical information is carried in the two FSK channels so that selective fading which disables one channel will not cause the compressor and expander to de-synchronize. Experiments on HF circuits have demonstrated that the use of in-band frequency diversity is necessary for the proper operation of the system.

The portions of the voice band needed to carry the control channel is pre-empted from the output voice band by a filter. The control channel is combined with the compressed (output) voice to form the audio input to the transmitter.

The placement of the control channel displaced from the edges of the voice band facilitates the retrofit of the system to different makes of HF radio by avoiding the poorly defined and widely variable transfer characteristics of SSB filters near the band edges.

3.2 RECEIVER

The audio output of the radio is split by filters into the three components consisting of the compressed audio and the two FSK control channels. The two FSK channels are demodulated independently and combined to form a single 75 bit per second data output used to control a microprocessor. The compressed voice input after appropriate filtering is sampled at 9600 samples per second. Gain changes are applied at the rate of 75 per second in the opposite direction to the compression process. In this way a system gain of unity is maintained. The microprocessor drives a D-A converter whose output, after filtering, forms the audio output.

4. TEST RESULTS

A series of tests was carried out between Ottawa, Ontario and Halifax, Nova Scotia and Ottawa, Ontario and Low, Quebec. The distance between Ottawa and Halifax is about 700 miles and the distance between Ottawa and Low is about 40 miles. Several frequencies between 3 and 11 MHz were used for the tests. Selective fading was frequently observed during the tests.

The test consisted of pairs of live voice transmissions, one of the pair with and one without Syncompex. The peak powers for the two transmissions were carefully equalized to ensure valid results. The time between the pairs of transmissions was kept to less than 2 seconds to minimize the effects of propagation and interference change. The operator at the receiver switched the Syncompex expander in and out as appropriate. The receiver audio was recorded in each case for subjective comparison.

Calibrated attenuators were inserted between the transmitter and the antenna to adjust the transmitted power of the link between pairs of transmissions. The purpose of the adjustment was to obtain some indication of the equivalent power advantage of the Syncompex system.

The test results can be summarized as follows:

1. A dramatic improvement in intelligibility and corresponding reduction of factors contributing to operator fatigue have been observed. The improvement is largely attributed to the silencing of noise by Syncompex during speech pauses. The listener feels more comfortable because he is more confident that he has not missed a low level syllable which could drastically change the meaning of a sentence. A secondary affect is related to the action of the AGC of the receiver when Syncompex is applied. The receiver gain does not fluctuate excessively during a speech pause because of the presence of the control channel. This leads to more natural sounding speech.

2. The improvement offered by Syncompex is dramatic for circuits that have marginal signal-to-noise ratio. When the signal is strong a slight impairment can be heard due to the loss of spectrum pre-empted for the control channel.

3. The Syncompex voice processing is judged to be equivalent to about a 10 db increase in transmitter power. This number is approximate because it is based on subjective judgement and is also dependent upon propagation conditions which varied during the tests. More precise evaluation would have to be made on statistical basis requiring more data than presently available. It should be noted that the HF-SSB radio used as a comparison reference does not incorporate more conventional voice processing techniques such as "clipping" used by radio amateurs.

4. Some tests were made with a Syncompex model having only a single control channel. (Ref. 4) The performance of this system was judged to be unsatisfactory because of its inability to withstand selective fading due to multipath. (Ref. 5) When fading conditions were encountered the single control channel was disabled intermittently causing sudden changes in the received audio level. Test subjects registered strong disapproval of this type of distortion because it sounded "unnatural". In some cases listening discomfort during selective fading was so acute that subjects reported that they preferred to sacrifice intelligibility. Since multipath conditions are frequently encountered, in-band frequency diversity to counteract selective fading is considered to be a necessary part of the design, and current versions of Syncompex include a dual control channel.

There is sufficient evidence that Syncompex can offer worthwhile improvement in performance to HF-SSB radio telephone systems. Further development is in progress to produce a design suitable for quantity production.

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