DATA TRANSMISSION RATE ENHANCEMENT FOR PACKAGE-SWITCHING RADIO TERMINAL

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Received October 5, 1979

Presented by Prof. Dr. S. CSIBI

1. Introduction

The system parameters of a radio network based on random time sharing or on packet switching, depend mainly on the radio channel data transmission capacity. This is obvious, because the random time sharing procedures utilize the high user activity (i. e. the ratio of the preparation time to the message packet transmission time). The packet transmission time 0.6 sec can be typical, assuming preparation time about 60 sec, packet of 80 characters and transmission rate e.g. 1200 bps. The values above together with the protocol of the channel determine either the number of users (terminals) capable of working simultaneously with determined average package delay time, or the average delay time in the case of determined numbers of terminals.

The number of serviceable terminals does not increase proportionally to the transmission rate but in higher rate. It can be shown, that based on a certain user behaviour not concerned here, assuming delay time of 1...2 sec, the number of serviceable terminals can be 8...13 on a channel of 1200 bps, and 140...170 on a channel of 9600 bps.

For some practical reasons the radio data transmission in random time sharing networks is practically realized by means of the widely used VHF radio equipments. Originally, these equipments were constructed for speech transmission and they are characterized first of all by the phase modulation of the audio-frequency range, limited to 3—3.5 kHz, using modulation factor of the value providing limited noise level in the adjacent channels, in the case of channel distance of 20 or 25 kHz (or 12.5 kHz for some cases).

These equipments should be made suitable either for data transmission in general or for transmission of message packets according to the special requirements. An obvious solution, causing only negligible modifications in the radio telephone equipment is placing a modem between the data source and the audio frequency input of the radio telephone. This method is called "subcarrier" method in order to be distinguished from the other method proposed in this work. There is a number of varieties of usual digital amplitude, frequency and phase modulation procedures to be used as modulation method. There were examinations published on the utilization of the available frequency range and on the highest rate possible. Summing up the measurement results of a radio telephone with audio-frequency bandwidth 5 kHz, for instance, it turns out that transmission 4.8k bps can be reached, using different methods and 9.6 kbps is available as well, under reasonable error rate and transmitted bandwidth, using special coding technique [2]. Increasing the speed, the equipment complexity and cost increase in the same manner, like those for modems in the telephone range.

For data transmission with radio telephone the *direct modulation* of the transmitted carrier using an appropriate method is another possibility. The given transmission bandwith can be utilized more effectively by means of linear modulation than using FM, or a given rate can be reached by means of simpler tools and with better characteristics than using the subcarrier method.

A version of binary phase modulator is presented, developed to be used with the equipment BRG FM 10—164, striving after the less modifications possible in the original equipment. The considerations in the planning, the direct modulator unit itself and later the results of the error rate measurements are presented briefly in the followings.

2. Realization of the direct modulation

The transmitter part of the realized system is illustrated by the block-scheme in Fig. 1 and the receiver part in Fig. 2. In the figures the original parts of the radio telephone equipment are shown above the dashed lines and the units providing the direct modulation or demodulation are shown under them. In the development of the modulator part a simple structure and realization were aimed at. A balanced multiplier was used as PSK modulator, the modulation itself was done on about 9 MHz and not on the final frequency of 160 MHz, using the signal of the basic oscillator of the radio telephone transmitter. The modulated signal was translated to the final frequency after suitable band limitation, by a frequency translator unit.

The data signal inputs to the modulator via a simple differential coder and shaping filter. It can be seen in the figure, that almost all the units of the radio telephone transmitter were used, except the audio frequency part and the FM modulator. There was no need for constructional modifications for the connections.

In planning the modulator part more effective solutions were searched for providing better system parameters. This way the coherent (synchron) demodulation, providing potentially the smaller error probability, was selected from the possible procedures to demodulate the PSK signal. The carrier generated from the modulated signal of suppressed carrier type is practically produced by means of PLL tracking system, for two contradictory requirements were to be met (i) the noise bandwidth of the carrier filter should be small, and simultaneously (ii) the signal should be found by the demodulator in the uncertainty region determined by the instability of the carriers of the radio

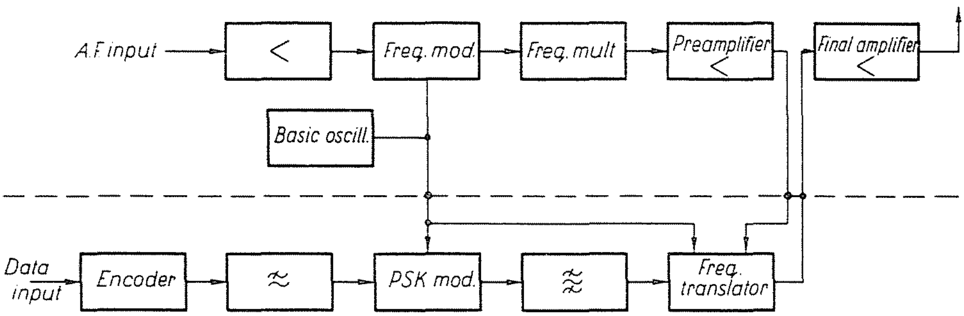

Fig. 1. The modulator and its connection to the radio telephone transmitter

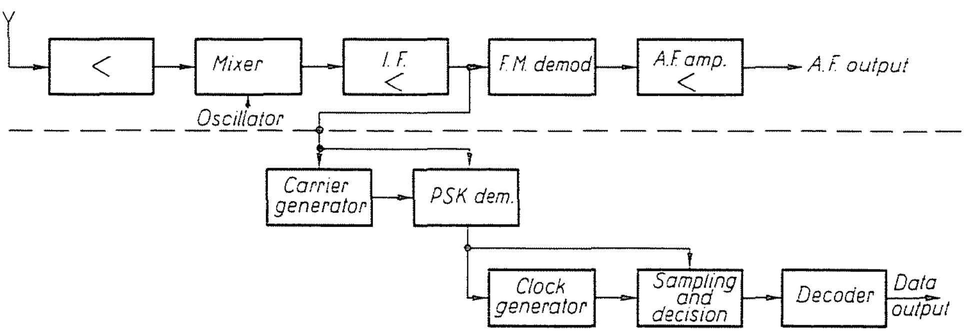

Fig. 2. The demodulator and connection to the radio telephone receiver

telephone. From the several suitable PLL systems fulfilling the requirements above, the Costas loop was selected, providing the carrier generation and the demodulation simultaneously. (The two units are shown separately in Fig. 2.)

The calculation and the realization of the Costas loop are not detailed here. The filters in the branches were realized, using RC integrating elements with cut off frequency approximately at 15 k Hz. The loop filter is a usual lag-lead type, having the pole and zero frequencies at 67 and 94 Hz respectively. The demodulator is a hibrid circuit, utilizing analogue and digital technique because of the operational frequency. The multipliers in the branches were realized with antivalence gates, and the inner multiplier was realized with analogue multiplier. The VCO is a standard quartz-stabilized integrated circuit. The phase shift is provided by using digital delay.

The sampler, the decision and the decoder circuits are the remainder significant circuits of the demodulator. They can be selected from commercial circuits, well proved in the practice. It is noted here at most, that several setting possibilities were provided according to the experimental purposes (e.g. the position of the sampling pulses).

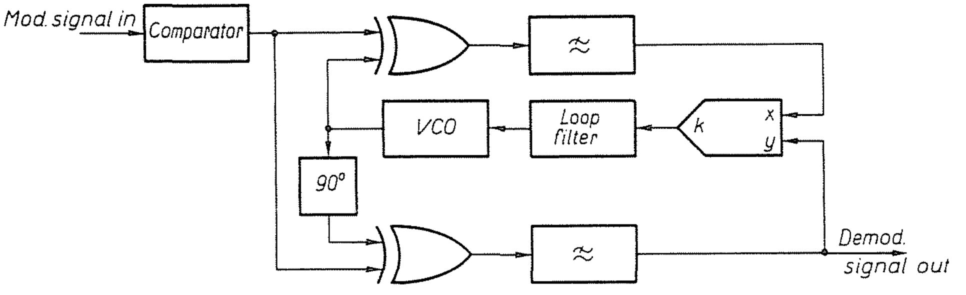

Fig. 3. The block scheme of the Costas loop

The demodulator unit is connected to the receiver of the radio telephone at the last stage of the i.f. amplifier, practically without constructional modification similarly to the modulator. The proper separation was performed by building in emitter followers.

In the precedings the modulation and demodulation tasks were considered. The corresponding units are connected to the terminal via an adapter. Our starting point was to provide standard interface V24 for the terminal, but the control lines perform special functions or are developed according to the special applications. E.g. the *Request to Send* signal (No. 105) turns on the power of the transmitter by means of a suitable switching circuit. (This way the carrier is emitted by the radio telephone only in the case of transmission.)

The signal *Clear to Send* (106) is produced by a circuit, following the appearance of the signal in the output of the transmitter. The received signal is detected by a high-speed circuit attached to the i.f. amplifier of the radio telephone receiver.

3. Measuring methods and results

The operation tests—in real radio telephone service—of the developed modulator-demodulator unit were not done until the time of the preparation of the present paper. This way only the laboratory test methods and measuring results are presented here.

The experiments were evaluated in two different arrangements (Fig. 4), looping a single modulator-demodulator.

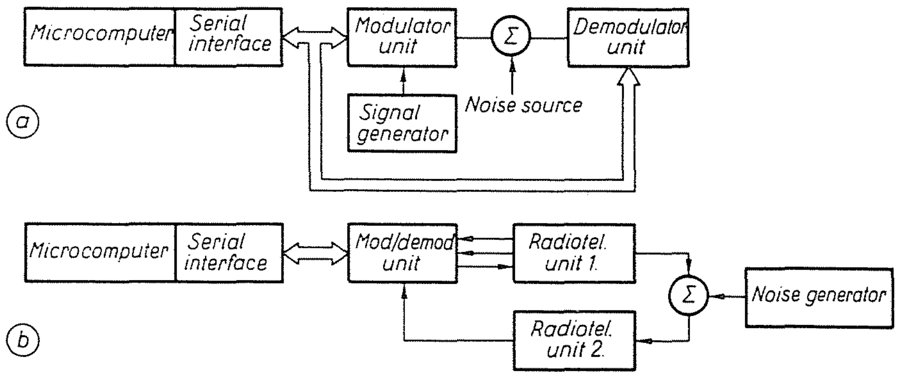

Fig. 4. Experimental assembly for measuring the error rate, a) Measurement at intermediate frequency; b) Measurement via radio telephone

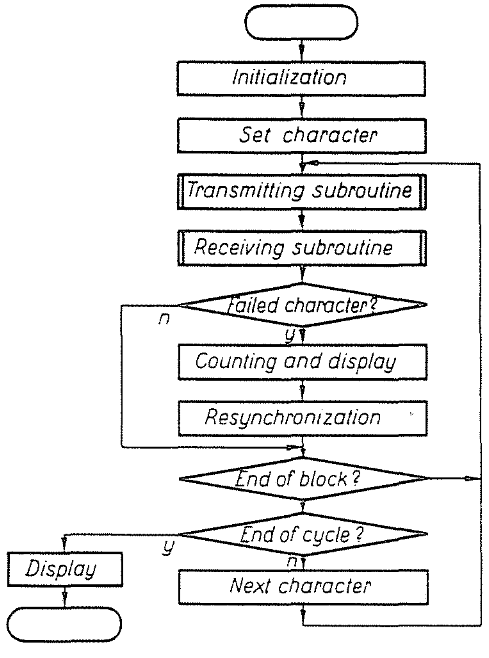

Fig. 5. Error rate measuring program flow-chart

An Intel 8080 based microcomputer built in our institute, was used in the measurements, acting as data source and error evaluator as well, using a program written for this purpose. The universal serial input-output unit of the microcomputer was suitable directly to be connected to the modulator-demodulator unit.

The circumstances of the planned application were imitated by the program, acting as data source and error rate counter as well and had several operation modes and setting and evaluating facilities. In one of the measuring modes (see the flow-chart in Fig. 5) blocks of 256 uniform characters, differing block by block were generated by the program, and the number of failed characters and bits and the total error number were displayed at the end of the cycle. In the case of failed character, the serial input-output unit is "resynchronized" by the program before the generation of the next character, i.e. byte synchron character is awaited, and the next character is sent after recognizing this.

This method was useful setting in the carrier and bit synchronizing circuits of the demodulator unit.

The error rate measurements were performed in an assembly shown in Fig. 4a using a noise source producing nearly white noise at 10.7 MHz with the bandwidth of approximately 30 kHz. Its value and the signal-to-noise ratio this way, was varied inserting an attenuator. Some preliminary measuring data are: The error ratio was better than 10^{-7} for signal-to-noise ratio 12 dB, and error ratio 2×10^{-4} was measured for signal-to-noise ratio 8 dB. These results are in good accordance with the known values for the demodulation of coherent PSK in the case of additive white Gaussian noise.

Summary

Enhancing the data transmission capacity of a radio channel is an effective way of increasing the number of terminals in random time sharing radio terminal networks. A "direct modulation" method was presented for this purpose, providing better utilization of the radio telephone channel than the known "subcarrier" methods. It can be stated, based on the experimental results, that transmission rate about 10 kbits/sec can be realized, using relatively simple equipments and without any strange signal shaping requirements.

The plannings of the circuits of the presented modulator-demodulator unit were done by József Molnár and László Szakács, electric engineers, graduated in 1978 and 1979, respectively as diploma theses. The authors thank for their valuable works.

References

1. Dallos Gy.—Szabó Cs.: Simulating Electronically the Performance of a Low-Rate Multiple Access Radio Channel. Proc. of the 6th Colloquium on Microwave Communications, Budapest, August 1978.
2. Zegers, L. E.—Dekker, C. B.: A Comparison of Digital Transmission Techniques for Standard FM Mobile Radio Sets. IEEE Trans. on Communications, March 1977.

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