

RECOMMENDATION ITU-R F.1112-1*

**DIGITIZED SPEECH TRANSMISSIONS FOR SYSTEMS
OPERATING BELOW ABOUT 30 MHz**

(Question ITU-R 164/9)

(1994-1995)

The ITU Radiocommunication Assembly,

considering

- a) that voice communications in the HF band use 3 kHz channels;
- b) that security is essential for some communications;
- c) that scrambling is the only means of obtaining a sufficient level of security;
- d) that the required level of security can easily be achieved using digitized speech technology;
- e) that there is therefore a need for speech signal coders (vocoders) associated with HF modems;
- f) that, for good quality HF channels, the maximum permissible bit rate is 4 800 bit/s;
- g) that interference and propagation effects such as fading cause an increase in the bit error ratio for digital communications, thereby calling for correction processes (error-correcting codes, interleaving),

recommends

- 1** that, for short-range communications (using ground waves), vocoders at 2 400 bit/s or 4 800 bit/s should be used;
- 2** that, for long-range communications (using sky waves), either vocoders at 2 400 bit/s or vocoders at 600-800 or 1 200 bit/s with error-correction coding should be used, according to the quality of the link;
- 3** that the digital radiotelephone systems used should comply with the general specifications set out in Annex 1 and the specific specifications contained in Annex 2, 3 or 4, according to the type of vocoder involved.

ANNEX 1

General outline of HF digital radiotelephone systems

An HF digital radiotelephone system comprises a conventional radiotelephone circuit, a speech signal coder (vocoder), an optional scrambling facility and an HF modem. Figure 1 represents the block diagram of such a system.

The transmitted speech signal is input to the voice coder, where it is analysed and converted into a bit stream. This bit stream, possibly after scrambling, is then applied to the modem, in which it is shaped for transmission in the telephone channel frequency band. The bit stream from the modem receiver output, if necessary after descrambling, is applied to the voice decoder, where the speech signal is restored.

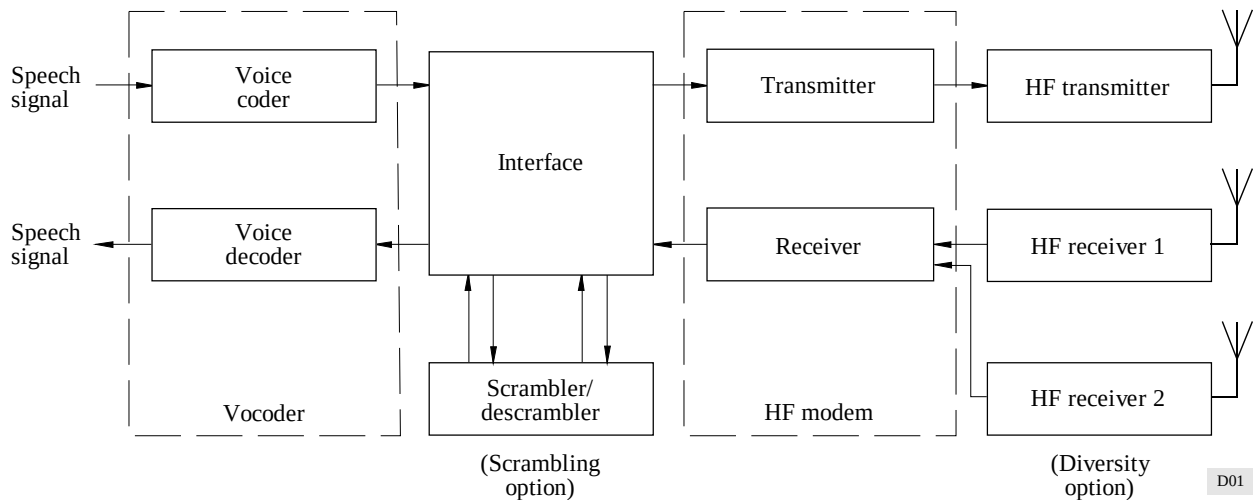
* This Recommendation should be brought to the attention of Radiocommunication Study Group 8.

For high error ratios with which vocoders at low bit rates (2 400 or 4 800 bit/s) cannot cope, vocoders at a very low bit rate (600-800 or 1 200 bit/s) with error-correction coding should be used.

Diversity reception can also be employed, by means of two receivers using space-diversity antennas, with subsequent processing in the receiving part of the modem.

There are many types of vocoder at bit rates $\leq 4\,800$ bit/s, in particular channel vocoders, orthogonal vocoders and linear predictive vocoders. Systems using those types of vocoder are described in Annexes 2, 3 and 4.

FIGURE 1
Block diagram of an HF digital radiotelephone



The three types of vocoder all have a wanted bit rate of 2 400 bit/s and a sound intelligibility greater than 90% under good transmission conditions. They are thus more or less equivalent in terms of quality at that bit rate.

All the vocoders are presented with the associated modem which in all three cases is of the “parallel modem” type (several sub-carriers independently modulated in the audio band).

For the channel vocoder and the orthogonal vocoder, the short period of the elementary symbol transmitted by the modem (13.3 ms and 8.33 ms, respectively) make them less resistant to severe HF propagation conditions, characterized by multipaths with a delay of a few milliseconds, which will cause significant inter-symbol interference.

For the LPC 10 vocoder, when the associated modem is the one described by STANAG 4197, the symbol period is 22.5 ms. This modem is thus less affected under such conditions.

Furthermore, in the LPC 10 vocoder certain important frame bits are protected by a powerful correction code (Golay code), with the result that the deterioration in performances as the bit-error ratio (BER) increases is more gradual than with the previous vocoders: sound intelligibility falls to 80% for a BER of 2%, as against 1% for the other vocoders.

Satisfactory operation also has to be guaranteed in poor conditions. This can easily be achieved with the LPC 10 vocoder by using simultaneously:

- data compression to reduce the wanted bit rate to 600/800/1 200 bit/s (at the cost of a drop in the vocoder’s intrinsic quality);
- insertion of redundancy, which increases resistance to errors.

With this arrangement, in the 800 bit/s version, the threshold of 80% sound intelligibility is not reached until the BER exceeds 5%.

ANNEX 2

Digital radiotelephone system using a channel vocoder**1 System description**

This system is composed of a conventional HF radiotelephone circuit and a digital voice terminal.

The speech input from a microphone is analysed and digitized by the channel vocoder. The digital signal is then applied through the interface to the scrambler where it is scrambled with randomizing signals. The scrambler output goes through the interface to the modulator and is converted to a voice-frequency signal by the FDM-D-QPSK method.

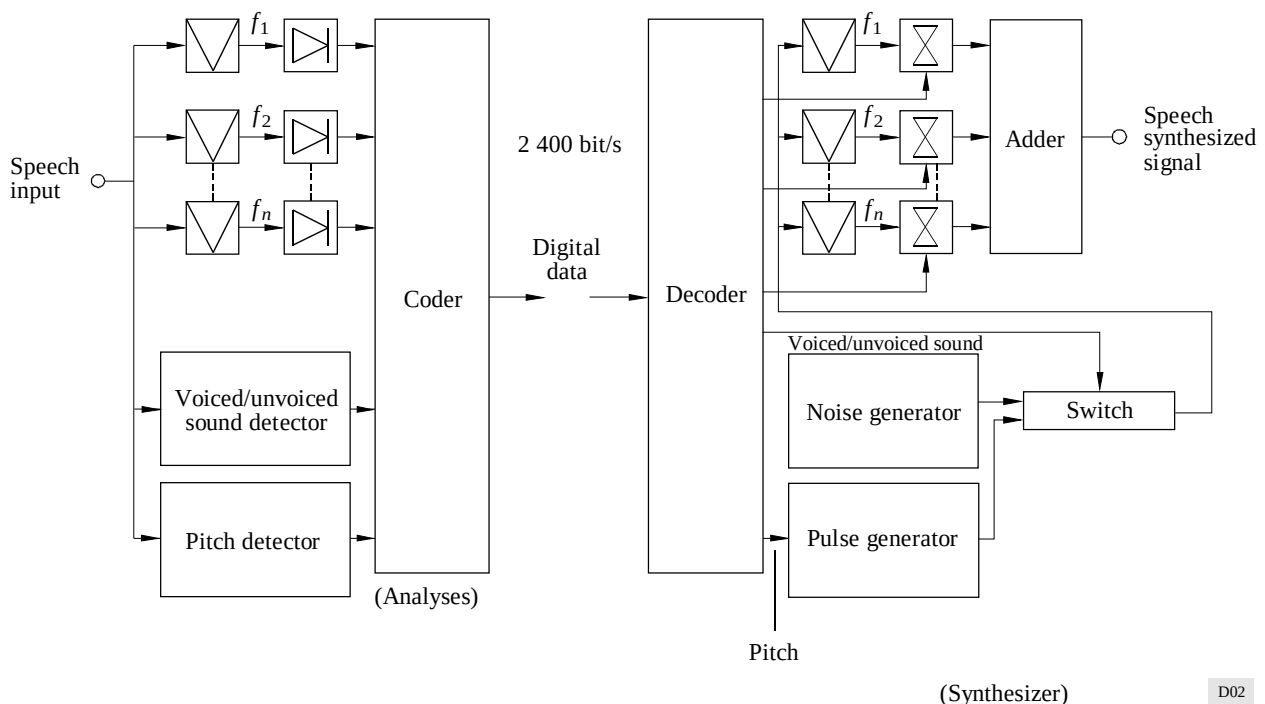
The audio signal output of the radio receiver is demodulated into digital codes by the FDM-D-QPSK demodulator. The digital code signal goes through the interface to the descrambler where it is translated into the original digital codes. These signals go through the interface to the channel vocoder to restore the speech signal, which is applied to a telephone receiver.

2 Channel vocoder**2.1 Theory**

The channel vocoder divides the speech-band signal of about 300 to 3000 Hz into a number of contiguous spectral bands, and measures the strength of each band. These measurements are coded and transmitted.

Figure 2 shows the channel vocoder theoretical block diagram. In the analyser section, a number of bandpass filters (BPF; filter-equivalent processing) are used to separate the speech bandwidth and to pick up the frequency component of each spectral band. The BPF outputs are measured to determine the level of each band. At the same time, “voiced” and “unvoiced” sounds and pitch frequency are detected. These parameters are sampled and quantified at the analyser, which formats a 2 400 bit/s coded signal for transmission.

FIGURE 2
Block diagram of channel vocoder



In the synthesizer section, a noise generator and an impulse generator excite the spectrum synthesizer, the noise generator for unvoiced sounds and the impulse generator for voiced sounds. The output frequency of the impulse generator is set almost equal to the pitch frequency. The output of the noise generator or the impulse generator is analysed by a BPF configuration similar to the analyser section. The levels of analysed spectral bands are multiplied and added to recover the speech signal.

2.2 Implementation

The digital voice terminal uses FFT (Fast Fourier Transform) for speech frequency spectrum analysis. The calculated spectrum is separated into a number of spectral bands whose bandwidth is equivalent to that of the BPF. The spectral bands or channels are averaged to determine the level of each spectral band. The frequency is detected by obtaining the maximum auto-correlation, while voicing-unvoicing detection is based on the level of the maximum values of the function. On the other hand, speech is synthesized by generating the impulse response of the BPFs with an FIR (Finite Impulse Response) digital filter. The result is multiplied by the output level of each spectral band. Finally, the waveforms of all spectral bands are added to get a speech signal.

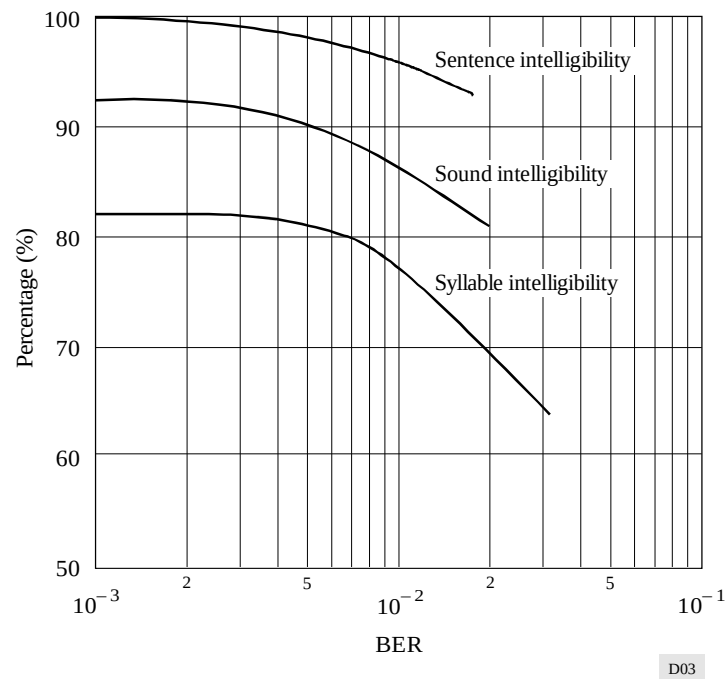
3 FDM-D-QPSK modem

The FDM-D-QPSK modem applied in this system is basically the same as described in Recommendation ITU-R F.763 (see Annex 1). The major characteristics are: data rate is 2 400 bit/s, 18 tones are used, 16 of these with a spacing of 110 Hz in the band 935-2 585 Hz being modulated in D-QPSK mode at 75 Bd signalling rate. A 605 Hz tone is used for the correction of end-to-end frequency errors. An 825 Hz tone is chosen for synchronization to avoid excessive loss at the band edge. Guard times between frames are introduced to combat multipath propagation and group-delay distortion.

4 Test results

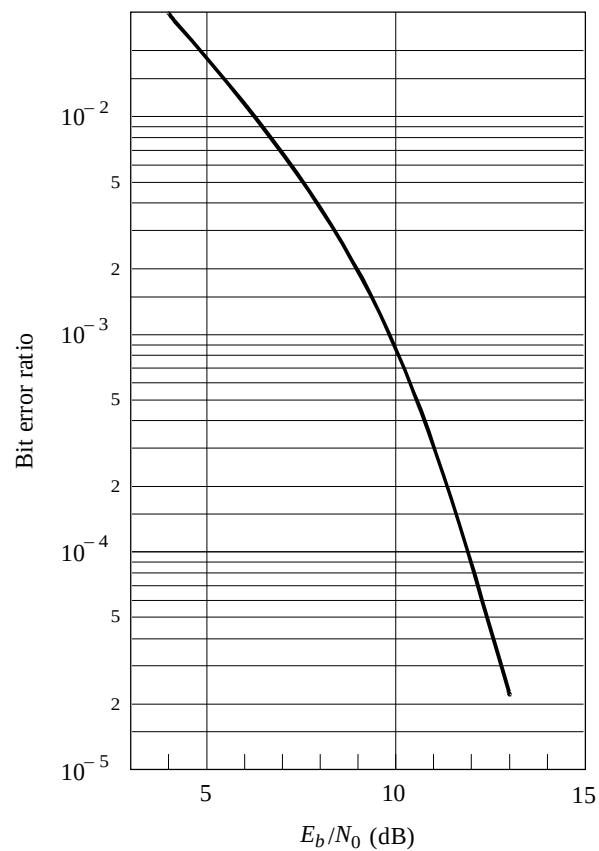
Figure 3 shows the characteristics of a channel vocoder in terms of sentence intelligibility, sound intelligibility and syllable intelligibility vs. BER. Figures 4 and 5 show the characteristics of FDM-D-QPSK modem in terms of BER vs. E_b/N_0 . Static characteristics are in Fig. 4 and characteristics with fading in Fig. 5.

FIGURE 3
Characteristics of a channel vocoder



D03

FIGURE 4
Static characteristics of FDM-D-QPSK modem



E_b : energy per bit

N_0 : noise power spectrum density

D04