

## 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.

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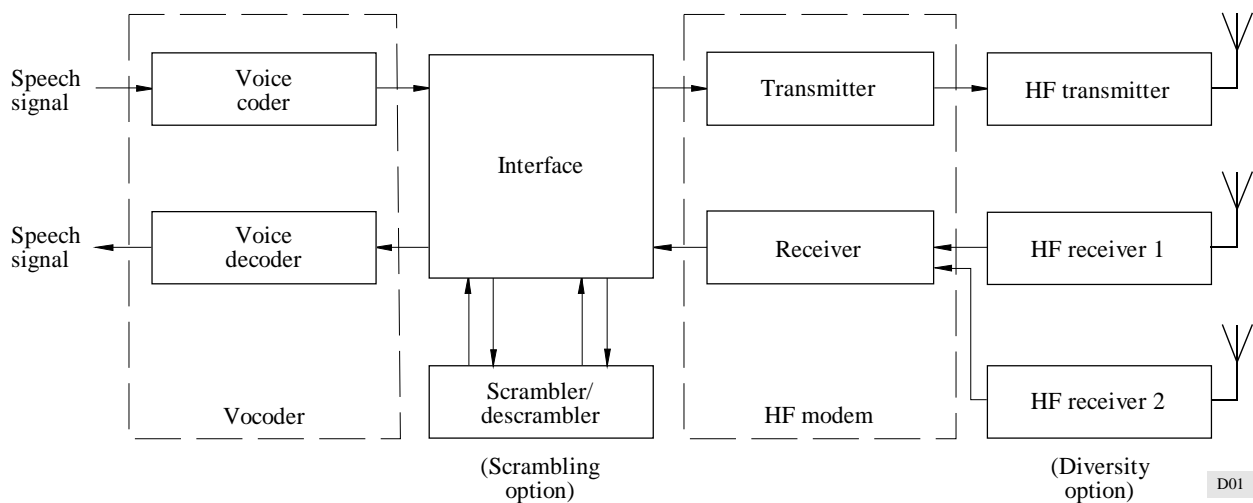
\* This Recommendation should be brought to the attention of Radiocommunication Study Group 8.

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.

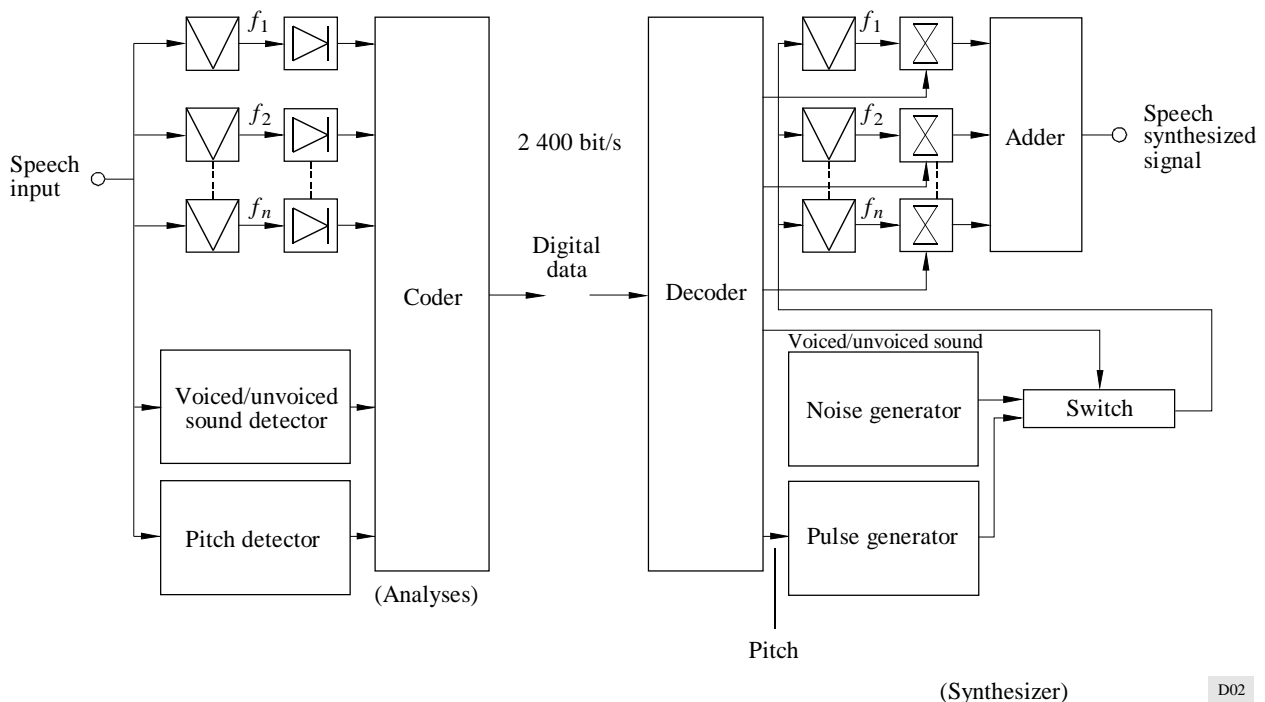
**2 Channel vocoder**

**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.

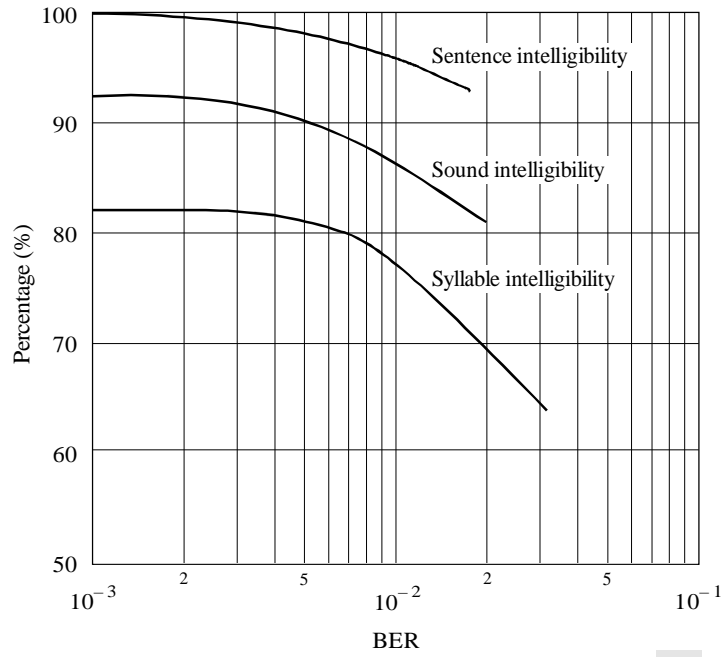
## 3 FDM-D-QPSK modem

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 2400 bit/s, 18 tones are used, 16 of these with a spacing of 110 Hz in the band 935-2585 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

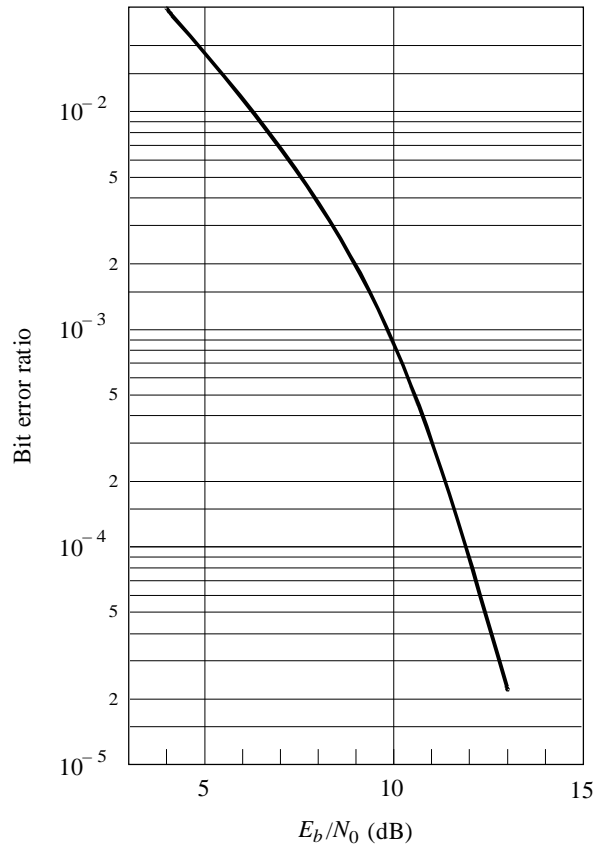
Figure 3 shows the characteristics of a channel vocoder in terms of sentence intelligibility, sound intelligibility and syllable intelligibility vs. BER. Figures 4 and 5 show the characteristics of FDM-D-QPSK modem in terms of BER vs.  $E_b/N_0$ . Static characteristics are in Fig. 4 and characteristics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

FIGURE 3  
 Characteristics of a channel vocoder



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FIGURE 4  
 Static characteristics of FDM-D-QPSK modem

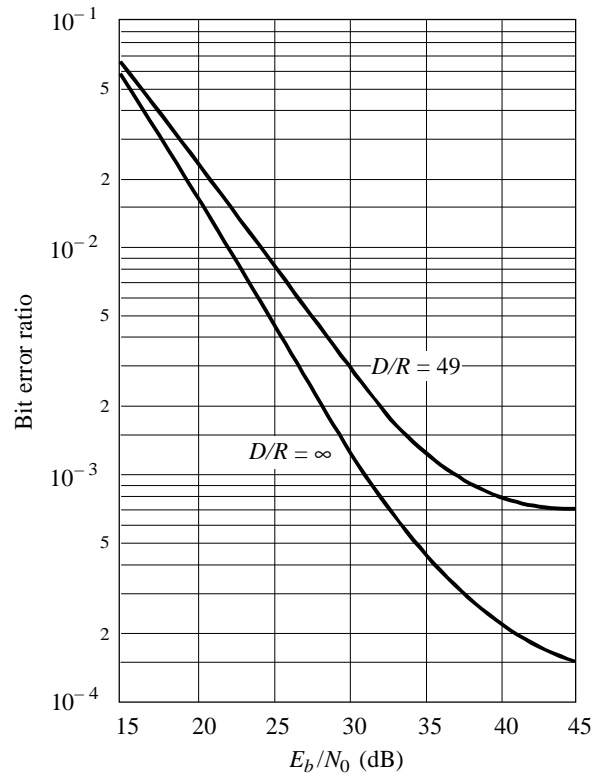


$E_b$ : energy per bit

$N_0$ : noise power spectrum density

D04

FIGURE 5  
 Characteristics of FDM-D-QPSK modem  
 with fading



Doppler frequency: 0.5 Hz  
 Time delay: 1 ms

$D/R$ : r.m.s power ratio between direct wave and reflected wave

D05

## ANNEX 3

### Digital radiotelephone system using an orthogonal vocoder

#### 1 System outline

Figure 1 represents a block diagram of a type of digital radiotelephone system. The circuit includes the conventional radio equipments of a main HF link: transmitters, receivers for dual space-diversity reception, antennas and trunk circuits.

The line terminal equipment consists of a vocoder for conversion of the speech signal into a 2 400 or 4 800 bit/s bit stream and a modem to communicate with the HF radio.

## 2 Vocoder

Operating tests were carried out on the digital radiotelephone circuit using two types of vocoder designed for operation at 2 400 and 4 800 bit/s. The block diagrams of the vocoders, both orthogonal, are shown in Figs. 6 and 7. The initial speech signal is applied to the spectrum analyser, which determines the values  $Y_k$  of the speech signal spectrum envelope at different frequencies.

The 4 800 bit/s vocoder takes 30 samples evenly distributed within each of three sections of the total speech signal frequency range. These samples are converted into 16 coefficients  $\gamma_j$  of spectrum envelope decomposition into an orthogonal series, which with a 60 Hz frame frequency constitute a 3 840 bit/s binary sequence. Simultaneously with the spectrum analysis, the value of the fundamental frequency period is extracted from the speech signal as well as the "fundamental frequency-noise" excitation characteristic, which are transmitted at a double frame frequency of 120 Hz in an 8-digit code, thus occupying 960 bit/s of the total bit stream.

At reception, the signals  $\gamma_j$  which control the orthogonal spectrum synthesizer, and the fundamental frequency and fundamental frequency-noise signals, which control the synthesizer excitation generator, are extracted from the received bit stream. The synthesizer excitation generator generates either a pulse group sequence at fundamental frequency and possessing a uniform spectrum, or a pseudo-random pulse sequence. The speech signal is synthesized at the synthesizer output.

The 2 400 bit/s vocoder (see Fig. 7) is orthogonal with non-linear conversion, in which the orthogonal series is the square root of the speech spectrum envelope. The samples  $Y_k$  of the spectrum envelope number 21 here, the distance between them increasing smoothly with frequency following the curve of equal articulations. These samples are subjected to square root extraction  $\sqrt{Y_k}$ , after which ten coefficients  $\gamma_j$  of square root decomposition from the spectrum envelope into the orthogonal series are determined. The values of these coefficients are transmitted every 20 ms in the total bit stream by 4-digit code combinations, also at 20 ms intervals.

At reception, the 2 400 bit/s stream is split up into the component signals, among which the fundamental frequency and fundamental frequency-noise signals control the synthesizer excitation generator, which is analogous to that in the 4 800 bit/s vocoder, and, in parallel, two spectrum synthesizers. The output of the first synthesizer is connected up to the excitation output of the second synthesizer, which both squares the spectrum to be synthesized and establishes a linear relation between the initial and the synthesized speech spectra.

FIGURE 6  
Block diagram of a 4 800 bit/s orthogonal vocoder

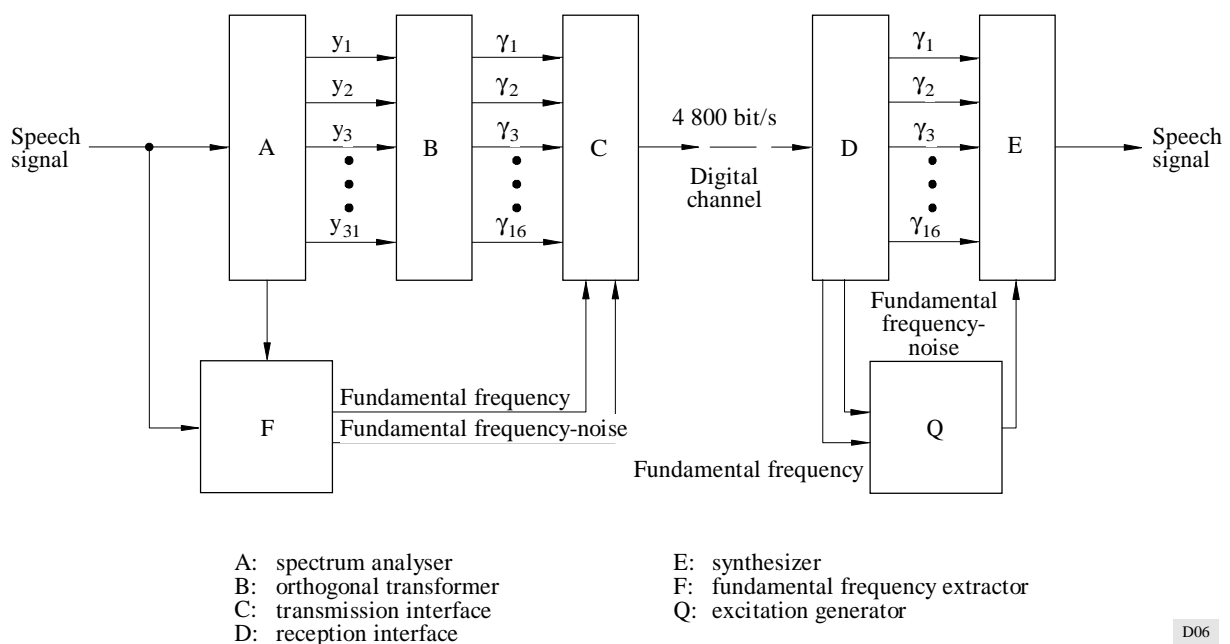
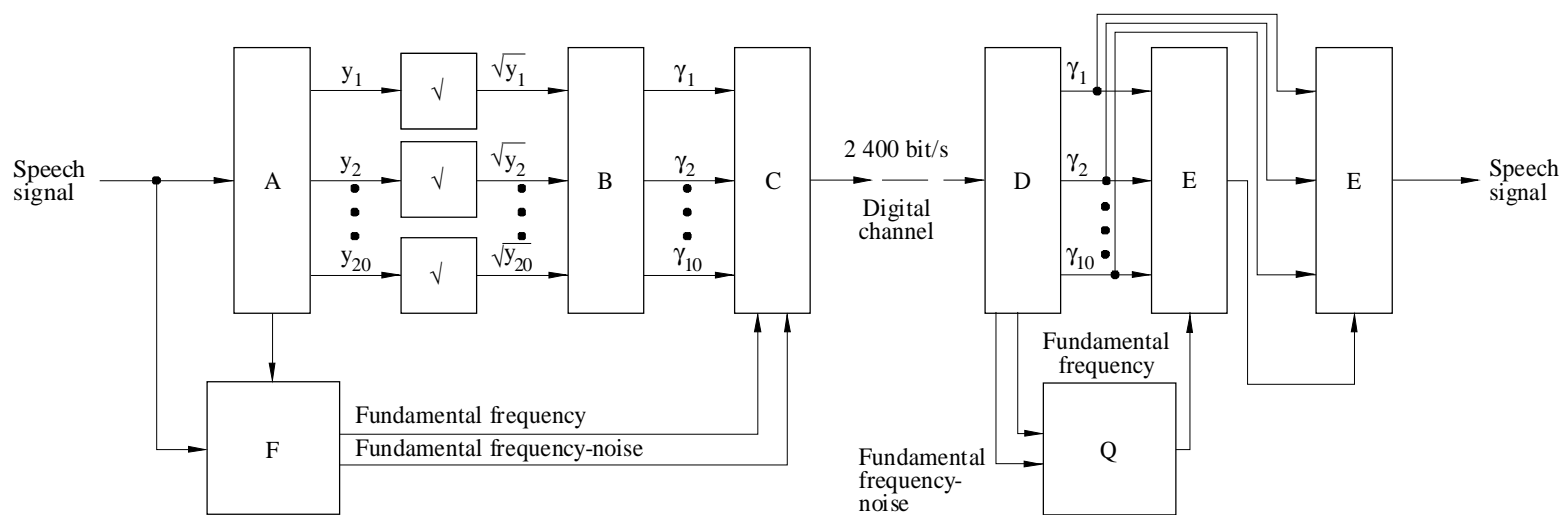


FIGURE 7  
Block diagram of a 2 400 bit/s orthogonal vocoder



A: spectrum analyser  
 B: orthogonal transformer  
 C: transmission interface  
 D: reception interface  
 E: synthesizer  
 F: fundamental frequency extractor  
 Q: excitation generator

D07



### 3 Modem

The modem is a 2-PSK multi-channel device using orthogonal signals. Its basic technical characteristics are as follows:

- sub-channel rate: 100 or 120 bit/s,
- number of channels: 20,
- channel frequency separation: 142 Hz,
- orthogonality interval: 1/142 s,
- length of protection interval: 1.29 or 2.29 ms,
- channel signal reception method: optimum non-coherent.

The modem is designed for the following rates:

- channel rate 4 800 bit/s: information rate 4 800 bit/s,
- channel rate 4 800 bit/s: information rate 2 400 bit/s (information doubling on sub-channel pairs with maximum frequency spacing),
- channel rate 2 400 bit/s: information rate 2 400 bit/s (1-PSK).

Furthermore, under 4 800 bit/s rate, the switchover to the 2 400 bit/s information rate is effected by Halley code.

The modem can also be used for reception from two space-diversity antennas.

## 4 Tests

### 4.1 Method

The operation of the digital radiotelephone circuit was studied over a period of several months at different times of year on latitude and meridian paths ranging from 1 500-3 000 km, and up to 10 000 km with repeaters.

Two methods were used for quality assessment:

- measurement of the intelligibility of the transmitted speech using articulation tables and of the reliability of bit stream transmission by measuring error rates in table transmission intervals;
- subjective statistical evaluation of speech quality made by subscribers after a lively exchange of conversation.

Altogether, more than 50 intelligibility and reliability measurements were carried out.

### 4.2 Analysis of the results

Trials have been carried out at different times of year and different times of day over a period of several months on main HF links ranging from 1 500 to 10 000 km in length. In these trials, articulation tables were used to measure the syllable intelligibility of the transmitted speech, while at the same time the reliability of the bit stream transmission was evaluated. Subjective statistical subscriber evaluations of the overall speech quality were also carried out.

The following conclusions may be drawn from an analysis of the test results:

- Error concentration in an HF link produces a wide dispersion of syllable intelligibility for an identical mean error ratio.
- Error concentration produces a greater effect on syllable intelligibility at a low error ratio (below  $10^{-3}$ ) than a uniform error distribution. This is due to the fact that, in the first case, the articulation table elements are as a rule affected not only by one but by several errors, with the result that they break down completely from the standpoint of sound recognition. However, subscriber evaluation of overall speech quality with error concentration is clearly more favourable than with an even error distribution at the same mean error density, since in connected speech a group of errors is perceived by the ear as an isolated single error, with the result that the overall impression is better.
- At a higher error ratio ( $5 \times 10^{-3}$  and more), grouped errors affect individual sections of speech, but leave fairly long sections intact, which means that a stable radiotelephone circuit can be sustained even at a mean error ratio of about  $10^{-1}$ .

- The use of dual reception with space-diversity antennas increases received syllable intelligibility by an average of 3-5% of syllables and appreciably improves overall speech quality.
- Speech transmitted by digital radiotelephony is more intelligible than speech transmitted over a conventional analogue HF circuit. The overall speech quality in digital radiotelephony is also greatly improved when measured by subjective tests, owing to the absence of the characteristic HF channel effects, such as selective fading and interference from other stations.
- In the operation of long digital radio links using repeaters, regeneration of the bit streams at the relay point enables a stable radiotelephone link to be sustained in conditions in which the normal methods, including Lincompex, do not ensure a satisfactory transmitted speech quality.
- Trials carried out on digital radiotelephone circuits using vocoders designed for operation at 2 400 bit/s and 4 800 bit/s have shown that, although the 4 800 bit/s vocoder produces better quality speech in a noise-free channel than the 2 400 bit/s unit, in real HF radio circuit conditions the latter, when combined with a modem operating at a channel rate of 4 800 bit/s with reduction to 2 400 bit/s, provides in most cases a better speech quality. In these operating conditions, digital speech telephony possesses clearly expressed threshold properties, and the quality of the transmitted speech with deteriorating propagation conditions remains high almost up until the moment when communications are completely cut off.
- In digital radiotelephony, the speech signal transmission factor is stable both overall and on the individual frequency components. Since long speech signal propagation delays render the echo signals much more audible, it is essential to fit an echo suppressor.

## ANNEX 4

### **Digital radiotelephone system using an LPC 10 linear predictive vocoder**

#### **1 System outline**

The system uses two vocoders at bit rates of 2 400 bit/s and 800 bit/s, respectively.

The first, at 2 400 bit/s, provides speech of sufficient quality to secure good intelligibility at a low bit rate.

The permissible line error ratio may be up to 1-2%, which is sufficient when the link is of good quality.

When the link quality deteriorates, i.e. when the error ratio at 2 400 bit/s exceeds 1.5%, the second vocoder at 800 bit/s is used. In this case, an error detection and correction procedure is employed which reduces the effective error ratio at 2 400 bit/s to a value compatible with use of the 800 bit/s vocoder, i.e. about 1%.

The second vocoder provides a slightly lower quality than the 2 400 bit/s vocoder; this may be considered as the necessary trade-off in order to continue communicating on a channel over which a link could not be set up with the initial vocoder.

The two vocoders include the following functions:

- analysis of the speech signal, in order to extract the set of parameters required to represent it adequately,
- quantification, which converts the digital values of those parameters into a bit stream for transmission,
- possible insertion of supplementary data (redundancy), in order to detect and correct transmission errors,
- de-quantification, to restore the original parameters,
- synthesis, which, on the basis of the received parameters, restores a speech signal designed to produce an acoustic impression as close as possible to that which would have been produced by the original signal, without attempting to reproduce the signal itself.

The analysis and synthesis functions are identical for both vocoders.

Only the quantification, redundancy insertion and de-quantification functions differ; while they both process a line bit rate at 2 400 bit/s, they are designed for a wanted bit rate of 2 400 or 800 bit/s, as appropriate.

## 2 Principle of LPC 10 vocoders

### 2.1 Modelling of the speech signal

First of all, the speech signal may be considered as (almost) stable over short periods; it may thus be segmented into *frames* of constant length (in this case 22.5 ms), where its characteristics are assumed not to change. Thus, all the necessary data to synthesize one or more speech frames are transmitted at regular intervals, independently of those which precede or follow them.

Then, as the available bit rate is very low, the speech signal has to be *modelled*, i.e. represented by a set of parameters as close as possible to physical reality, in other words to the phenomena responsible for producing it.

To that end, two cases are considered according to whether the signal is periodic (voiced) or non-periodic (non-voiced).

For periodic sounds, which correspond to vowels, the speech signal is assumed to be obtained by spectral shaping (filtering) of a periodic excitation, which in fact constitutes a fairly accurate reflection of physical reality, where the acoustic vibrations of the vocal cords are propagated in the vocal tract (larynx, mouth, etc.) which acts as a filter whose transfer function depends on the vowel pronounced.

There are two sorts of non-periodic sounds:

- stable or semi-stable sounds (such as the sibilants, like the letter “S”);
- transitory sounds (such as the plosives, like the letter “P”).

These two cases only differ in terms of their duration and the rate of change of the sound level.

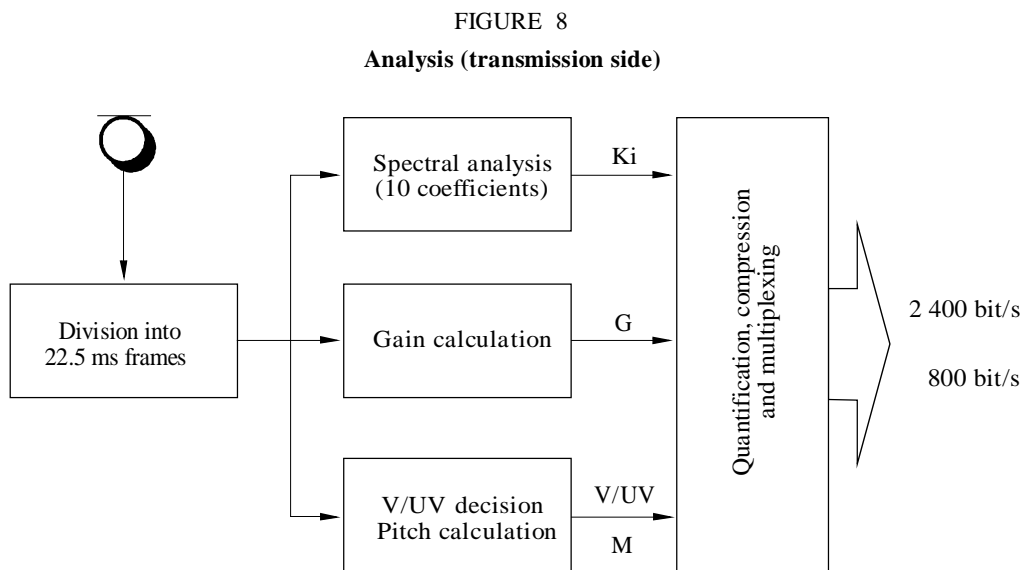
Both are represented as the output of a filter to which the input is a more or less rapidly changing random excitation (white noise); again, this reflects the real situation, where this type of unstructured sound is produced by sudden turbulences or occlusions in the vocal tract.

Finally, the overall speech signal is still represented as a variable-level and variable-type (periodic or aperiodic) excitation, filtered so as to reproduce as accurately as possible the frequency spectrum of the original signal.

The filter calculation is based on the principles of predictive analysis, whence the name of the process - linear protective coding (LPC).

### 2.2 Analysis

The analysis processes extract from the speech signal the parameters required to model it. They are shown in the block diagram in Fig. 8.



- Ki: reflection coefficients
- G: gain for the synthetic signal
- V: voiced sound (vowels)
- UV: un-voiced sound (consonants)
- M: pitch value

The first parameter to be determined is the type of signal (voiced or non-voiced). It is evaluated on the basis of periodicity criteria, which vary according to the case concerned. In general, one calculates the long-term auto-correlation of the speech signal (i.e. whether it is the same at regular intervals), thereby providing a first estimate. This estimate is then refined on the basis of other criteria, such as instantaneous level, short-term auto-correlation, ratio between signal energy at low frequencies and at high frequencies.

The result is a *voicing indicator*, which will be 1 if the signal is considered to be voiced, and otherwise 0.

When the signal is considered to be voiced, its period has to be estimated in order to reflect its height. This period, called *pitch*, is also evaluated in general on the basis of long-term auto-correlation; the value of pitch is the lapse of time after which the same speech signal is repeated.

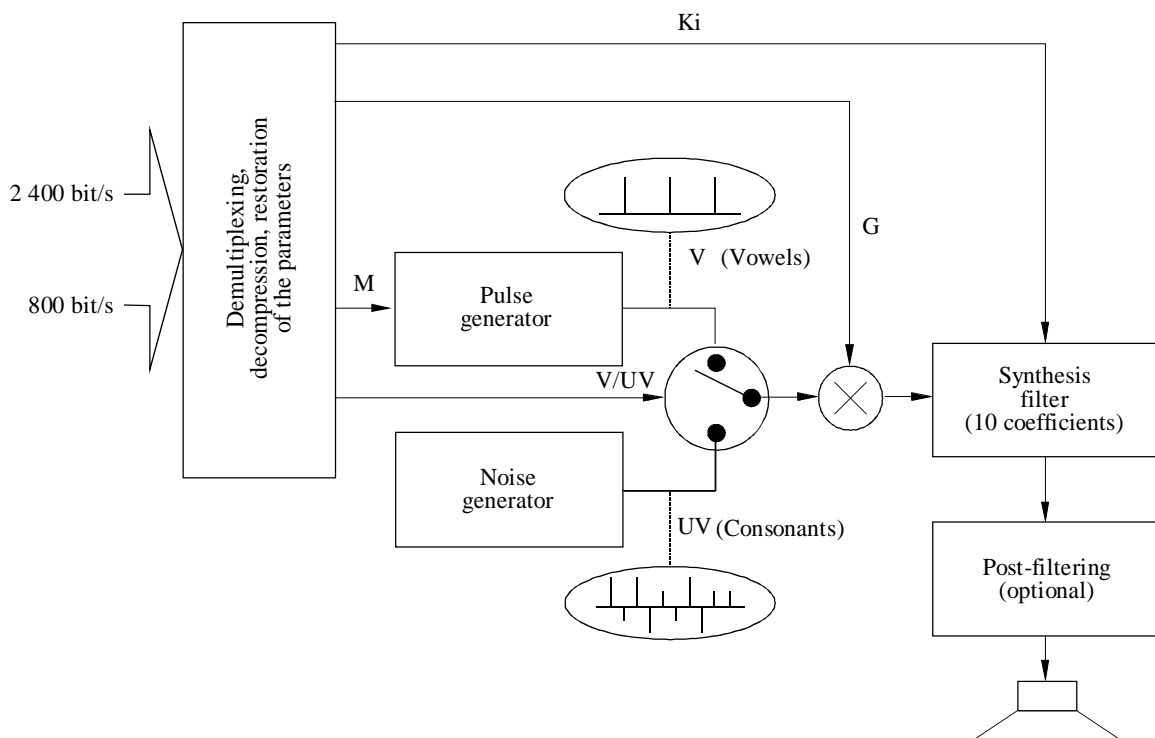
The filter, whose frequency response is as close as possible to the speech signal spectrum, is calculated on the basis of short-term auto-correlation coefficients from which the filter coefficients are deduced by conventional signal processing methods. The result of this analysis is a set of ten reflection coefficients (whence the name LPC 10) which sufficiently faithfully describe the cross-sectional variations in the vocal tract which originally filtered (coloured) the initially spectrally neutral excitation.

Finally, for each frame, the level of the speech signal is evaluated in order to control the gain of the synthesizer on the synthesis side.

### 2.3 Synthesis

The algorithms employed to synthesize the speech signal reflect the assumed speech production model. They are represented in Fig. 9.

FIGURE 9  
Synthesis (reception side)



$K_i$ : reflection coefficients  
 $G$ : gain for the synthetic signal  
 $V$ : voiced sound (vowels)  
 $UV$ : un-voiced sound (consonants)  
 $M$ : pitch value

They include, in succession:

- a noise generator, used for unvoiced sounds;
- a periodic signal generator, to which the pitch is provided, for voiced sounds;
- a switch allowing selection of either generator according to the type of speech signal to be produced in the current frame;
- a filter of order 10, which filters the excitation selected in order to give it its colour; it is at this level that the distinction between the different vowels and the different consonants is made;
- a gain control system, which gives the synthetic signal the right volume;
- optionally, a “post-filtering” system, designed to mask certain imperfections in the synthesizer and to make the synthesized signal more pleasant to the human ear.

### 3 Vocoder at 2 400 bit/s

For the 2 400 bit/s vocoder, the different parameters are quantified independently, frame by frame.

The pitch and voicing are quantified jointly over 7 bits, the level over 5 bits, and the ten filter coefficients over 41 bits for voiced frames, and a little less for unvoiced frames (the remaining bits being used to test the link quality).

After addition of a housekeeping bit (synchronization), frames of 54 bits are obtained.

The vocoder is covered by a NATO standardization agreement (STANAG 4198) and a United States Federal Standard (Fed. Std. 1015).

### 4 Vocoder at 800 bit/s

#### 4.1 Principles used to reduce the bit rate

The reduction in bit rate by a factor of three hinges on the observation that not only is there a correlation between the values of the different parameters from one frame to the next but also that the necessary quantification accuracy varies according to the context.

For example, it is rare for the observed level on consecutive frames to vary to any large extent, with the result that it is advantageous to code the level in blocks.

In addition, if the sound is stable, the predictive filter is too, with the result that there is no point in providing a new predictive filter for each frame. Although the available bit rate is reduced by a factor of three, it is still possible to quantify accurately a common filter for, say, three consecutive frames. On the other hand, in a transitory portion, the very concept of frequency spectrum tends to disappear, in which case one can quantify roughly one filter per frame in the knowledge that the representation error will be masked by the sudden variation in sound level.

Consequently, the basic principle used to reduce the bit rate consists in grouping speech frames together in packets of, say, three and encoding each packet of parameters as a block.

#### 4.2 Quantification procedures

Each data frame comprises 54 bits representing three speech frames. One of the envisaged quantification methods, for which property rights have been taken out, is described below.

A total of 10 bits are used to quantify both the pitch and the voicing. Account is taken of the fact that there cannot be more than one voiced → unvoiced transition (or *vice versa*) in a packet of three frames and that, if there is more than one voiced frame, the pitches of adjacent frames are similar; it is sufficient to transmit a reference value and an increment.

For the level, 9 bits are used, namely:

- 4 bits to define a reference level common to the three frames,
- 5 bits to describe (reading from a table called the dictionary) changes in level over the three frames.

Finally, for the predictive filter, a total of 35 bits are used, in two fields.

The first field, of 32 bits, is considered as relating to:

- a single filter common to the three frames, or
- a filter common to two successive frames and an increment to obtain the remaining filter, or
- two filters.

The second field, of 3 bits, is responsible for describing the coding scheme selected, from among eight possibilities.

The best coding scheme is selected on the analysis side in such a way as to minimize the spectral distance weighted by the signal level in each of the frames; a low-level frame is less well handled than a neighbouring frame of a higher level.

### 4.3 Correction coding

The correction code used to step up from 800 to 2 400 bit/s is a block code comprising 54 wanted information bits for each packet of  $3 \times 54 = 162$  bits transmitted.

One may select, for example, a simplified Reed-Solomon code with symbols of 6 bits, comprising 27 symbols in all for nine wanted symbols. This code, RS(27,9), allows a total of nine errors to be corrected, i.e. a *maximum* proportion of 33% of errored symbols or an *average* proportion in the order of 20%.

## 5 Resistance to transmission errors

Resistance to transmission errors is obtained by comparing the respective intelligibilities of the two vocoders as a function of the error ratio at 2 400 bit/s, which is the bit rate transmitted by the modem over the HF channel. That intelligibility is plotted in Fig. 10.

It will be seen that intelligibility is slightly lower at 800 bit/s when there is no transmission error.

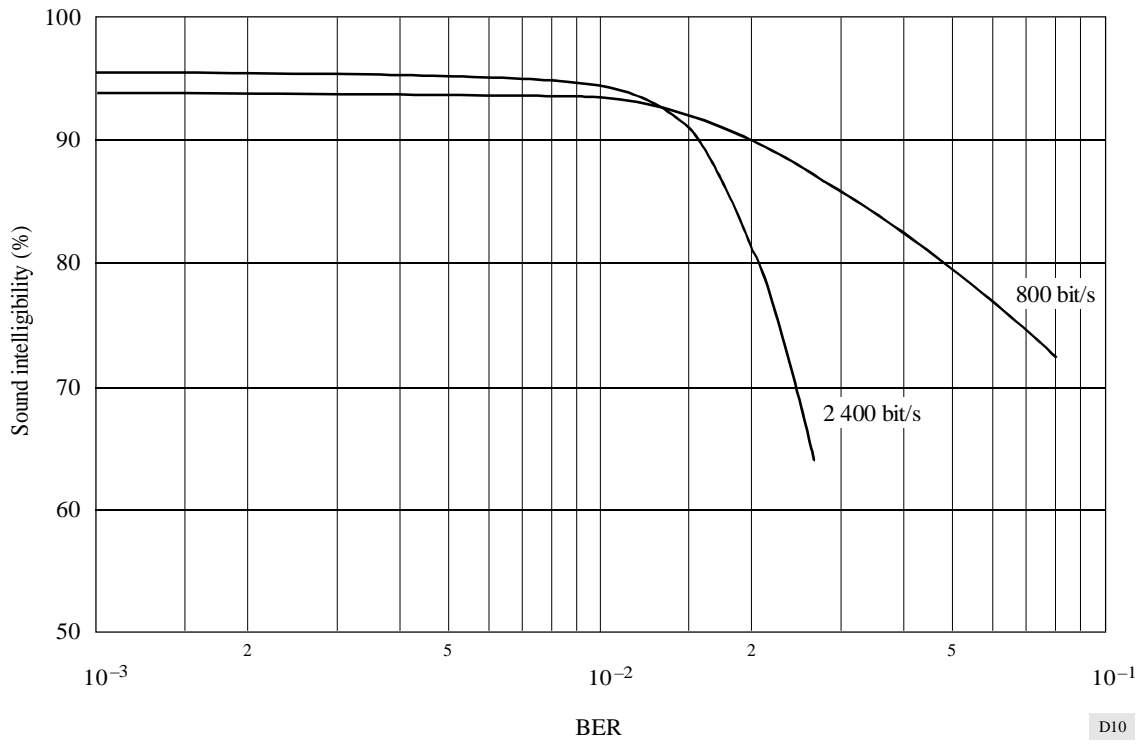
However, as the error ratio increases it deteriorates more slowly than at 2 400 bit/s, for the following two reasons:

- the correction code significantly reduces the error ratio when the changeover is made from 2 400 to 800 bit/s;
- when the error ratio is high at 2 400 bit/s, this results in deleted frames at 800 bit/s and deleted frames have less of an effect than errored frames.

## 6 Modem

Transmission for both vocoders is carried out at a bit rate of 2 400 bit/s, using a standard modem, for example the modem described in Recommendation ITU-R F.763 or the one described by STANAG 4197.

FIGURE 10  
Intelligibility of LPC 10 vocoders



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