

## RECOMMENDATION ITU-R SM.1600

**Technical identification of digital signals**

(2002)

The ITU Radiocommunication Assembly,

*considering*

- a) that the use of radio grows steadily;
- b) that digital signals are being widely used;
- c) that an increasingly large number of devices can be used without benefit of a licence or certification process, preventing any administration to identify their emissions;
- d) that the interference complaints involving digital emissions are often difficult to resolve;
- e) that prior identification often is an essential prerequisite to any measurement on digital signals with complex waveforms as used in digital broadcasting (digital audio broadcasting (DAB), digital video broadcasting (DVB)) and cellular radiotelephony (time division multiple access (TDMA), code-division multiple access (CDMA)),

*recommends*

that digital signals should be identified in the following order:

- 1 general identification process as described in Annex 1;
- 2 identification based on the fundamental components of the signal as explained in Annex 2;
- 3 identification based on the waveform characteristics of the signal as described in Annex 3.

## ANNEX 1

**General identification process****1 Preamble**

Concerning the identification of digital signals, it is interesting to conduct the process of analysis as follows:

- a) Preliminary identification of known or standardized signals using intercorrelation method should be applied.

Standard emissions having specific characteristics (synchronization words, training sequences such as those included in many TDMA protocols, short spreading codes in

certain CDMA protocols, etc.) can be detected by intercorrelation techniques which provide at the same time a preliminary identification of these signals. This is particularly interesting for spectrum monitoring of waveforms used in UHF terrestrial digital radiocommunications (GSM, IS 95, DECT, etc.) and in satellite communications (GLOBALSTAR, ICO, IRIDIUM), while it is very simple design and offers a great flexibility of use. The main elements that have to be searched by intercorrelation method are the following:

- preamble;
- guard time;
- word of synchronization, training sequences;
- codes;

and, in the event of failure of the preliminary identification.

- b) Identification of the signal according to its waveform characteristics should be applied.

Concerning the identification of the transmission signals according to their design features, which finds its place in the event of failure or of impossibility of a preliminary identification, it is recommended to conduct the following process of analysis:

- preliminary detection of situations of multi-emission, separation of the emissions when it is possible;
- discrimination of analogue, non-(C)OFDM ((coded) orthogonal division multiplex) digital signals and (C)OFDM digital signals;
- identification of the characteristics of transmission of the analogue signals;
- identification of the characteristics of transmission of non-(C)OFDM digital signals;
- identification of the characteristics of transmission of (C)OFDM digital signals.

The main elements that have to be analysed are the following:

- recognition of digital/analogue signal;
- channelling and bandwidth of the signal;
- spectrum shape, shape of the temporal signal, shape of signal in the time/frequency domain (continuous/burst signal, fixed frequency signal/spread spectrum signal, frequency spreading characteristics of the burst signals: slot duration, rate of transitions between slots, etc.);
- response of the signal to various transforms (higher order moments, zero crossing, etc.);
- type of modulation (linear/non-linear; etc.);
- characteristics of modulation according to its type (number of symbols; transitions between symbols; modulation depth, etc.).

These different signals parameters can be exploited for identification according to methods recommended hereafter.

## 2 Signal capture and digitizing set-up to carry out signal identification

To this effect, the use of a digital system of reception of adjustable bandwidth is recommended for translation to  $I/Q$  or to intermediate frequency, signal digitization and recording of signal samples.

- a) For the narrow-band signals likely to be identified in real time, one recommends to use input filters with fixed centre frequency and bandwidth,  $B$ , followed by narrower filters with adjustable centre frequency bandwidth, in order to obtain bandwidths well centred on the studied emissions; magnitude of suitable  $B$  values is:
  - telegraphic or telephone bandwidth emissions: 100 Hz to 3, 4 kHz;
  - emission of medium bandwidth: 15 to 45 kHz.
- b) For the broadband signals, rather subject to identification in differed time, one recommends the following magnitudes for the values of signal capture filters, while allowing post-processing with more selective filters:
  - $B = 300$  kHz (for most radiocommunication signals of today);
  - $B = 2$  MHz (mainly for DAB and IS 95 signals);
  - $B = 8$  to 10 MHz (for the broader emissions such as DVB, UMTS and IMT-2000 signals, etc.).
- c) Higher bandwidths require digitizers and processors which are more expensive, as well as high bit rate transmission means in case of remote controlled stations, while corresponding to a lesser quantity of emissions. Therefore, monitoring stations could be set-up in a first stage with the lower bandwidths only.

For an acceptable system, it is recommended to use the following the material components:

- an analogue or a digital receiver with fine adjustable centre frequency and performing high dynamic, high gain control magnitude (50 to 60 dB for the magnitude of the gain control are the typical recommended values);
- filters, baseband converters, analogue to digital converters and recorder providing:
  - more than 8 bits magnitude for each signal sample (12 bits magnitude is the typical value recommended);
  - sampling rates providing more than 4 samples for each digital modulation symbol;
  - storage depth providing a recorded signal duration of a few milliseconds for wideband signals and of a few seconds for narrow-band signals.

## ANNEX 2

## Preliminary identification by intercorrelation techniques with a test signal and by other correlation techniques

Intercorrelation methods are based on the following theoretical consideration:

### 1 A brief theoretical overview of the correlation techniques

These intercorrelation techniques use known elements in the signal to be measured that allow intercorrelation calculation for detection, synchronization, time and Doppler compensation.

In practice, these known signals are modulated signals inside the waveform whose symbol stream is given inside norms, system specifications etc.:

- synchronization word in many standardized continuous like waveforms (such as frequency division multiplexing (FDM));
- training sequences in TDMA standardized waveforms;
- PILOT codes or synchronization words in standardized CDMA or TDMA/CDMA waveforms, etc.

NOTE 1 – To face an unknown signal to be processed, it is possible to use a previous recorded signal as known signal or to point out such signals by autocorrelation calculation. In all cases, important simplifications occur if the recording conditions of the known signal are known, especially the relative speed or Doppler shift of the signal source.

Considering a known signal,  $z(t)$ , included inside an emitted signal,  $s(t)$ , and considering the related received signal,  $x(t)$ , and an integration duration,  $IT$ , the intercorrelation and the autocorrelation function are given in the following form:

$$\Gamma_{x,z}(\tau) = \frac{1}{IT} \cdot \left[ \int_{\zeta=\tau}^{\zeta=\tau+IT} x(\tau+\zeta) \cdot z^*(\zeta) d\zeta \right] \quad \text{and} \quad \Gamma_{x,x}(\tau) = \frac{1}{IT} \cdot \left[ \int_{\zeta=\tau}^{\zeta=\tau+IT} x(\tau+\zeta) \cdot x^*(\zeta) d\zeta \right]$$

and if  $x(\tau+\zeta) \equiv z(\zeta)$  during  $[\tau, \tau+IT]$ , we have  $\Gamma_{x,z}(\tau) = \Gamma_{z,z}(\tau) = \frac{1}{IT} \cdot \left[ \int_{\zeta=\tau}^{\zeta=\tau+IT} |z(\zeta)|^2 d\zeta \right]$  that is

the mean power of the known signal  $z$  during  $\tau$  and  $\tau+IT$ .

In these expressions:

$\cdot$ : symbol for multiplication

$z^*(\zeta)$ : complex conjugate of the signal  $z(\zeta)$ , and  $|z(\zeta)|$  is the modulus of the signal  $z(\zeta)$ : if we write  $z(\zeta) = a(\zeta) + i \cdot b(\zeta)$  with  $a(\zeta)$ : real part of  $z(\zeta)$ ,  $b(\zeta)$ : imaginary part of  $z(\zeta)$ , and  $i^2 = -1$ , then  $z^*(\zeta) = a(\zeta) - i \cdot b(\zeta)$  and  $|z(\zeta)|^2 = (a(\zeta))^2 + (b(\zeta))^2$

$x^*(\zeta)$ : complex conjugate of the signal  $x(\zeta)$ , with the same definition as above

$\tau$ : delay time.

If  $x(\tau + \zeta)$  is equal to the known signal  $z(\zeta)$  during the interval  $[\tau, \tau + IT]$ , this corresponds to the max value of  $\Gamma_{s,z}(t)$ , equal to the mean power of the received signal  $z(t)$  during  $IT$ , that gives the time of arrival of the signal  $z(t)$  included in the received signal  $x(t)$ .

In practice, the received signal  $x(t)$  contains the emitted signal  $s(t)$  that is filtered by the propagation losses  $h$  and  $x(t)$  contains a noisy contribution  $n(t)$  and sometimes an external interference contribution  $j_{ext}(t)$ . So that the complete received signal  $x$  has an expression such as  $x(t) = h*s(t) + n(t) + j_{ext}(t)$ . In the later expression, the symbol  $*$  represents the convolution product and the symbol  $h$  represents the channel impulse response of the propagation filter that includes atmospheric attenuation, ground effects, multipaths, etc. Practical models of  $h$  are given for example in cellular standards. Note that the most simple approximation of the propagation is to consider that the channel impulse response  $h$  is equal to a mean attenuation coefficient  $h_0$  to be applied to the emitted signal  $s(t)$ , so that  $h*s(t) = h_0 \cdot s(t)$ : this later approximation will be considered in the following for simplification. At the synchronization time (or arrival time)  $t_{SYNC}$ , the emitted signal  $s(t)$  is equal to the known signal  $z(t)$  or includes the known signal  $z(t)$  (we will consider here after that  $s(t) = z(t) + (s(t) - z(t))$  where  $z(t)$  is the known signal inside  $s(t)$  and  $s(t) - z(t)$  the unknown part of signal  $s(t)$ ). So, at synchronization time  $t_{SYNC}$ , the total received signal  $x(t)$  as such a form:

$$x(t_{SYNC}) = h_0 \cdot z(t_{SYNC}) + n(t_{SYNC}) + j(t_{SYNC})$$

In these later expressions, we note  $j(t_{SYNC}) = [h_0 \cdot (s(t_{SYNC}) - z(t_{SYNC})) + j_{ext}(t_{SYNC})]$  and  $j(t)$  represents the total interference contribution related to the unknown part  $s(t) - z(t)$  of the emitted signal  $s(t)$  and to the external interference  $j_{ext}(t)$ . From the later expression and by introducing the density of power spectrum related to signal  $z$  and to signal  $n + j$  and the bandwidth, BW, of signal  $z$ , and by considering a convenient filtering and sampling of the signal bandwidth, BW, the  $S/N$  ratio at intercorrelation output and at input are related by the following theoretical expression, where  $C_z$  is the power of the signal  $z$ , and  $N + J$  is the total power of noise + jamming in the bandwidth BW:

$$\left( \frac{C_z}{N + J} \right)_{output} = \left( \frac{C_z}{N + J} \right)_{input} \cdot BW \cdot IT$$

This later expression shows that the inter-correlation processing provides a receiving gain proportional to the product  $(BW \cdot IT)$ . For sampled signals that respect the Nyquist criteria,  $BW \cdot IT = L_{sample}$ : number of signal sample during the integration duration  $IT$ , so that the intercorrelation gain  $10 \log_{10}(BW \cdot IT) = 10 \log_{10}(L_{sample})$  is directly given by the number of signal samples. For example  $10^6$  signal samples assuming the Nyquist criteria would provide an intercorrelation gain equal to 60 dB. Under this hypothesis, Figs. 1 and 2 point out the interest of intercorrelation technique for detection of low digital signal processor (DSP) signals (simulated case).

## 2 Practical implementation of the intercorrelation technique

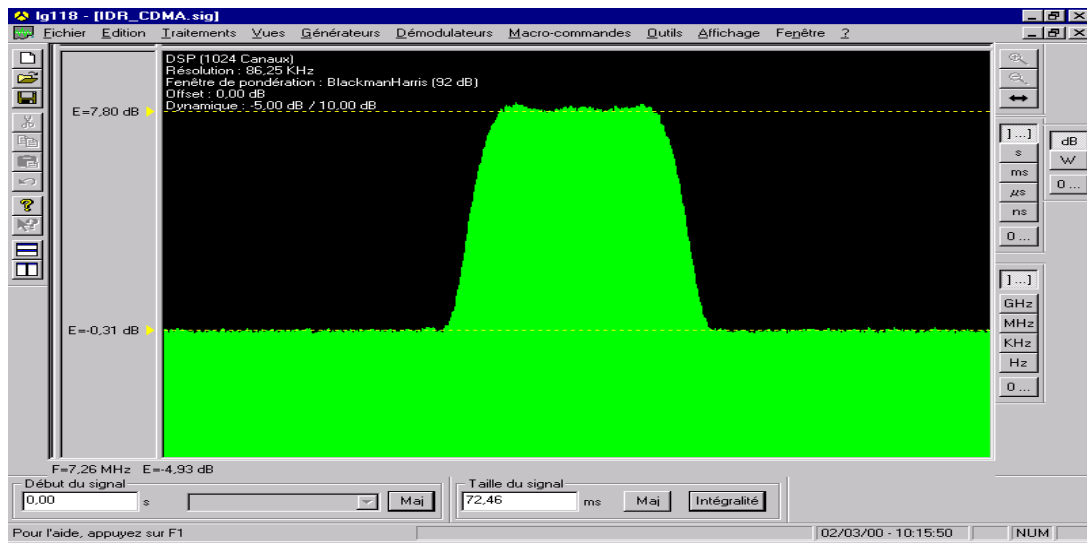
The practical implementation of these techniques use slides windows for determination of the time of arrival of the signal, and Doppler compensation techniques to take account of the movement of the signal source. Generally the practical methods use two steps:

*Step 1:* first, estimate the Doppler frequency error and the time synchronization instant;

*Step 2:* second, correct the Doppler frequency error and optimize detection and source separation.

FIGURE 1

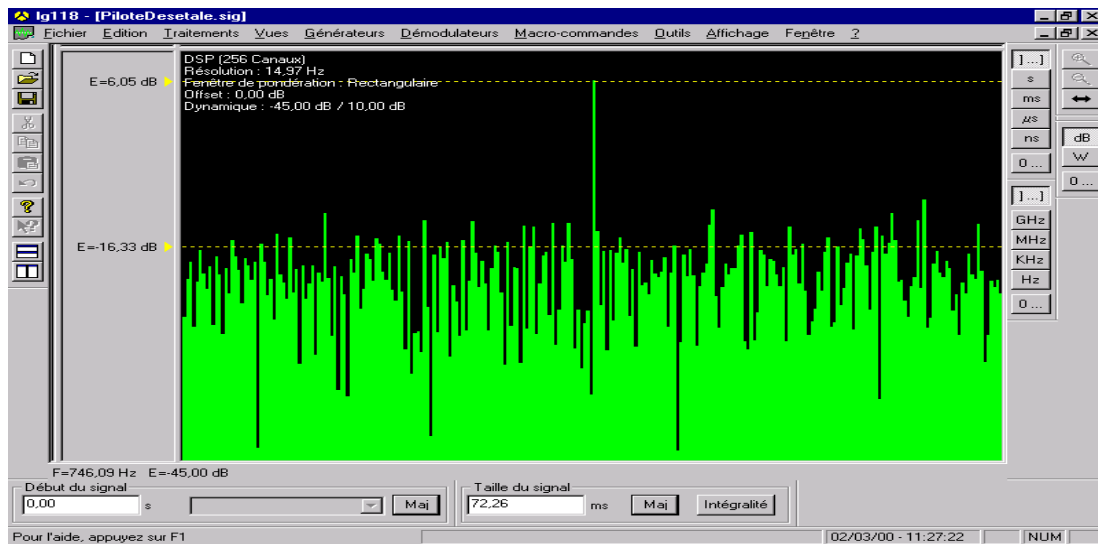
Spectrum of one dominant signal  $S_1$  + Gaussian noise + a secondary lower CDMA signal  $S_2$ .  
 $S_1$  is 8 dB greater than the noise floor  $N$  and is clearly pointed out by the spectrum view,  
 $S_2$  is 13 dB lower than the noise floor and not visible at spectrum view



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FIGURE 2

View of the intercorrelation of the previous signal (including  $S_1$ ,  $S_2$  and  $N$ ) with a pilot code included in the CDMA signal  $S_2$ . It clearly appears a pitch related to  $S_2$ , which is about 20 dB above the noise floor in the case



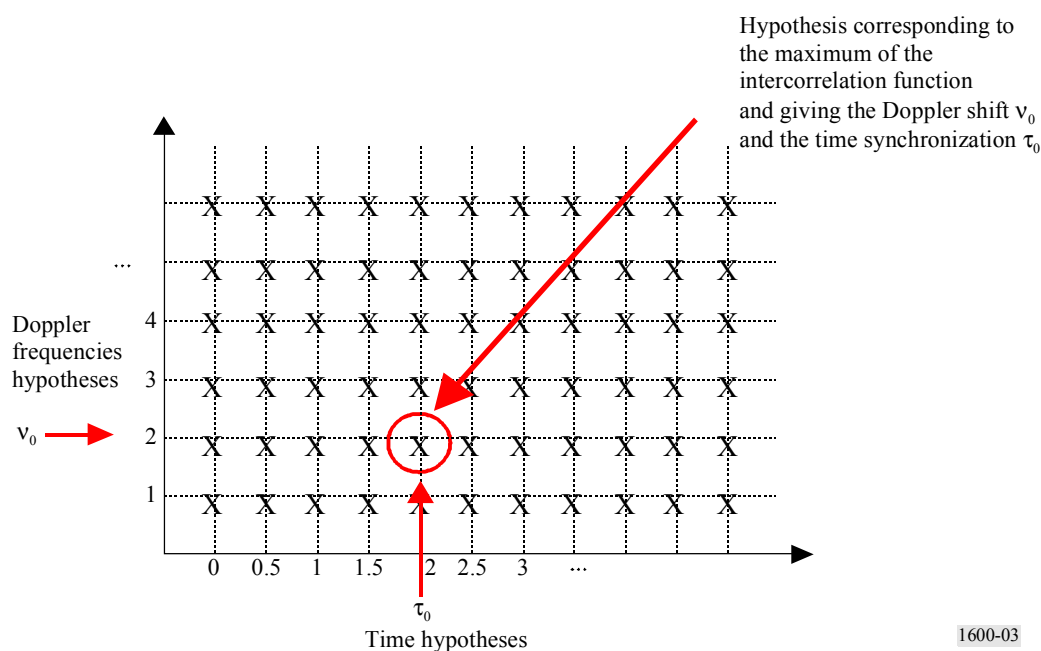
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## 2.1 Description of the time and Doppler synchronization

The principle of time and Doppler synchronization is to perform intercorrelation of the signal with the known signal for different time and Doppler shift hypothesis. Following this way the synchronization instant and Doppler frequency are jointly estimated by the point of the time frequency plan that realizes the maximum of the intercorrelation peak.

FIGURE 3

Searching the hypotheses in the frequency/time plan that optimizes the intercorrelation pitch



This processing may be achieved in a simplified manner by fast Fourier transforms (FFTs) techniques. In order to decrease the complexity of the FFTs, the coherent despreading may be performed on successive slices of signal, in such a way, that the carrier Doppler has a limited effect on a given signal slice.

## 2.2 Optimizing detection and source separation

For this, it is necessary:

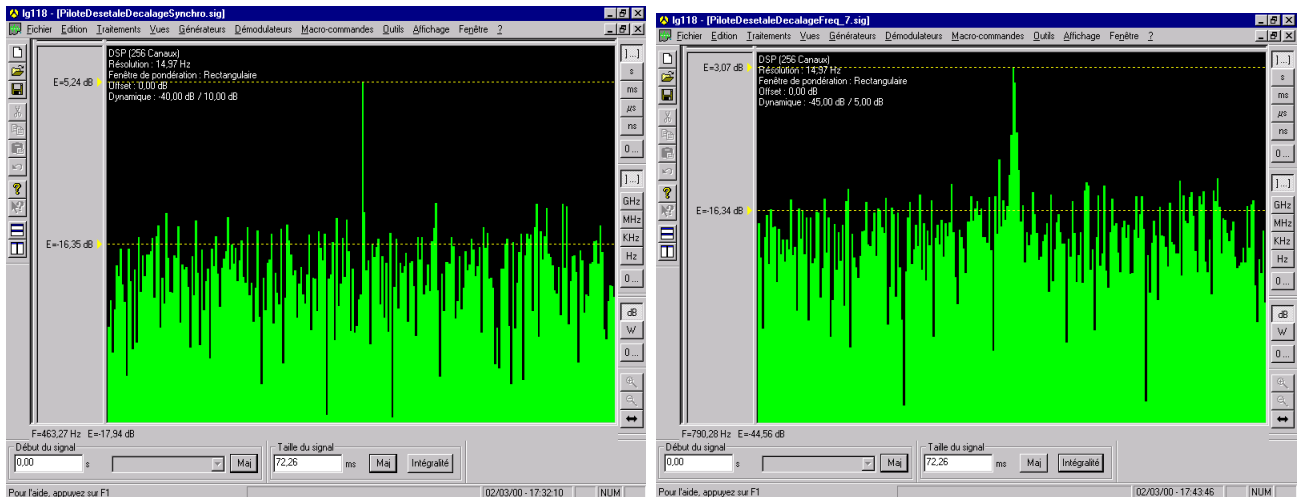
- to optimize the choice of the integration time, IT, this may be achieved by determination of the  $C/I$  and the  $S/N$  of the signal to be detected;
- to evaluate the precision of the determination of the synchronization time and Doppler, this may be achieved by several methods based on adaptive sampling, interpolation in the time/frequency plane;
- to estimate the degradation due to residual time and Doppler synchronization mismatch. Usually, the time and Doppler synchronization performances are expected to be spoiled by at least 4 factors whose importance is related to the integration duration, IT, the nature the  $C/I$  and the  $C/N$  ratios of the signal, and to the practical implementation of the synchronization algorithm:
  - FFT estimator with Doppler frequency error;
  - the residual frequency Doppler after compensation;
  - the timing mismatch of the reception in case of Nyquist filtering (sampling error);
  - the interference caused by other signals related to the intercorrelation matrix.

As an illustration, considering for example the CDMA signal  $S_2$  of Fig. 1 which detection is achieved in an ideal case at Fig. 2, Fig. 4 shows the detection of the same signal  $S_2$  under the same condition except of:

- a time mismatch of  $T_c/(2 \cdot N_e) = T_c/4$  ( $T_c$ : chip duration) at Fig. 4a);
- a frequency mismatch of 8 Hz for Doppler synchronization in Fig. 4b).

FIGURE 4

Detection with time mismatch and Doppler mismatch

a) Time mismatch of  $T_c/4$ 

b) Doppler mismatch of 8 Hz

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The degradation here is less than 1 dB for the pitch at Fig. 4a) but it is about 4 dB at Fig. 4b).

### 3 Application of the intercorrelation techniques for preliminary recognition and identification of transmission signals

Therefore, the tools for identification must be able to provide the following functions:

**3.1** Digital reception system of adjustable bandwidth for frequency translation and recording of the sampled signal allowing to apply subsequently real time and differed time processes.

#### 3.2 Generation of test signals – Constitution of a database

One recommends to be able to carry out the generation of test signals:

- by simulation (generation of a transmission signal with control of the bit stream, the modulation, the rate and the carrier), and by using software and hardware signal generators;
- by extraction from files of existing signals (cut and paste on a digitized signal, coming for example from a signal generator or from previously recorded real signals);

and this without limits other than the data-processing capacities of the machine used (great numbers of signal samples can be necessary for these techniques according to the sampling rate and the



length of the test signals), in order to get maximum benefit of the future increase in computer memory size and central processing unit (CPU).

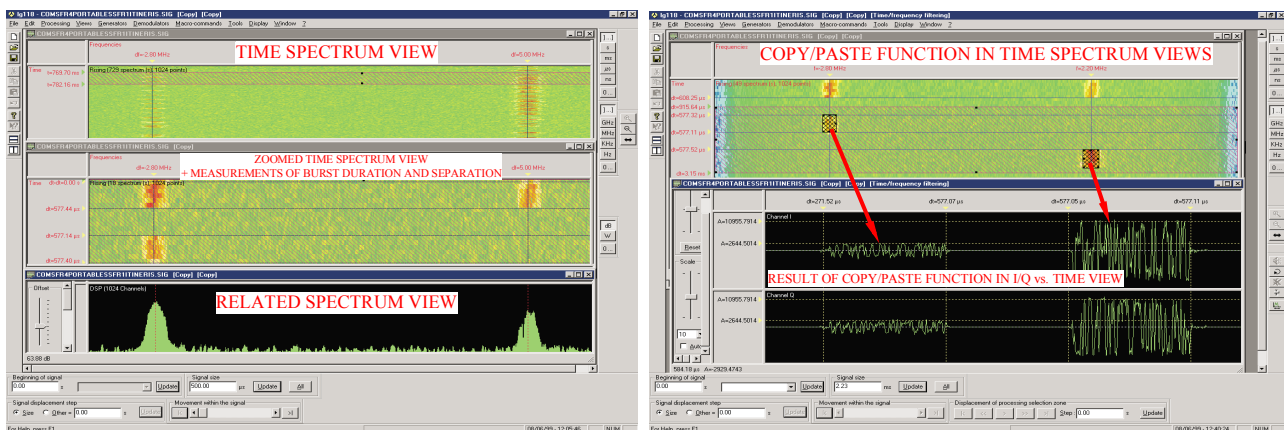
### 3.3 Preliminary analysis to help the user to choose the suitable test functions

For that, one particularly recommends the use of cursors or grids adapted to the common standard waveforms and applied to time, spectrum and time/frequency displays. This allows a rapid determination of the carrier frequency, channelling, bandwidth, rate and duration of signals, as well in the case of multicarrier signals, as in the case of burst signals, under conditions of moderate to heavy spectral occupation (cellular network for example).

These measurements (with functions of zoom, copy and paste, etc.), frequently lead to an immediate recognition of the nature of the signal as shown in Fig. 5 (GSM signal recognized by its burst duration of 577  $\mu\text{s}$  and its 200 kHz channelling).

FIGURE 5

**Time spectrum representation and panoramic display of a GSM burst signal, with associated zoom, copy/paste and measurements functions**



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In Fig. 5, recognition of the GSM signal is confirmed by measurements: channel spacing is a multiple of 200 kHz (in this case 600 kHz), burst duration is 577  $\mu\text{s}$ , etc.

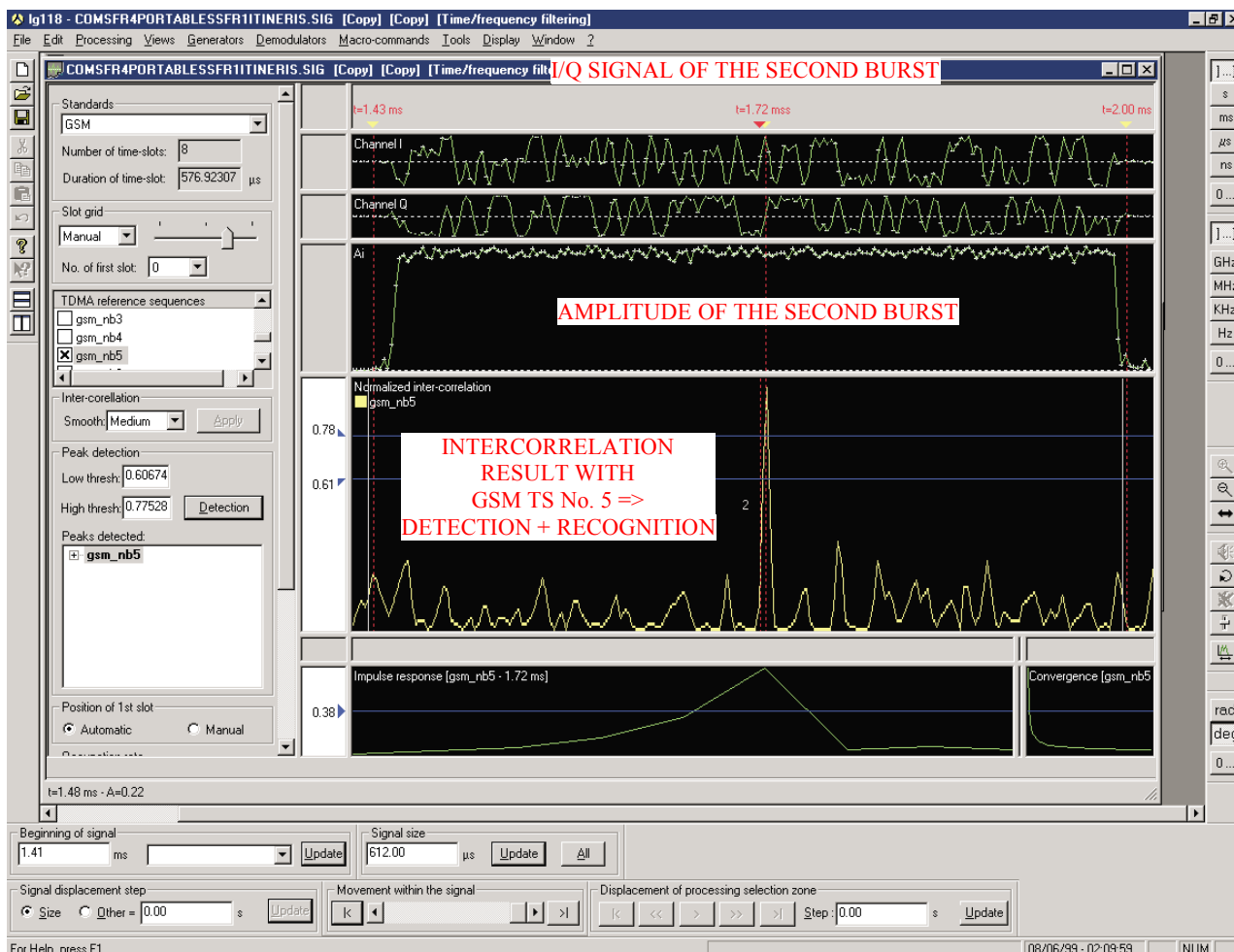
### 3.4 Intercorrelation with test signals and search functions

The peak value of the intercorrelation of the analysed signal with a test signal makes it possible to detect the presence of the test signal inside the analysed signal. Thus, the intercorrelation of a signal with the reference sequences makes it possible to specify its technical identity.

For example, in the case of a GSM signal of Figs. 5 and 6, which is easily recognized by time/frequency characteristics (577  $\mu\text{s}$  burst, periodicity multiple of 8, channelling multiple of 200 kHz), the systematic intercorrelation with training sequences considered here as test signals makes it possible to specify it is a traffic channel using the training sequence No. 5 (see Fig. 6). In the same manner, another channel of traffic, bursts of synchronization, bursts of frequency correction, and stuffing bursts can be identified in broadcasted channels or traffic of cellular networks.

FIGURE 6

Recognition of the technical identity of a GSM burst; a traffic burst is identified: its intercorrelation with the midamble sequence No. 5 is nearly one



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This method applies also to words of synchronization, PILOT codes or codes of synchronization employed in the CDMA protocols so that their waveforms can thus be dispread and identified.

Additional information can easily be obtained such as estimation of the channel impulse response, the verification of demodulation quality and information on the occupancy per time slot in the case of a TDMA signal (see Fig. 6).

### 3.5 Systematization and automation of the test signal search process

The process of preliminary identification of the signals is very effective with standard protocols having words or sequences of synchronization (this is common for many TDMA waveforms), PILOT codes (this is common for many CDMA waveforms), etc., but it may require heavy computing power. One recommends the use of accelerated algorithms making the process highly systematic and automated, and this, in order to process the test signals by family of standards, such as:

- PMR mobile radio systems, cellular systems, terrestrial or satellite systems based on a TDMA protocol (including TETRA, GSM, D-AMPS, PDC, DECT, IRIDIUM, ICO, etc.);

- PMR mobile radio systems, cellular systems, terrestrial or satellite systems based on a CDMA protocol (including IS 95, UMTS, IMT-2000, GLOBALSTAR, etc.).

For that, it is desirable:

- to carry out simultaneously several tests by intercorrelation (test by family of standards, i.e. on all the known signals of a given standard, for example TETRA, GSM, IS 136, IS 95, etc.);
- to associate to intercorrelation techniques suitable displays together with a graphic assistance to highlight the characteristic patterns of the signals, using functions of peak search, recurrence search, etc., such as those shown in Fig. 2.

## ANNEX 3

### Waveform characteristics verification

#### 1 Introduction

Concerning the identification of the transmission signals according to their modulation characteristics features, it is recommended to conduct the following process of analysis:

#### 2 Detection of multi-emissions

##### 2.1 Practical importance of the detection of multi-emissions

To obtain a good measurement and to increase chances to identify a signal according to its design features, the operator is tempted to choose a wide filter for scanning, typically several kHz in HF and a few tens or hundreds of kHz at VHF/UHF, in which several emissions may coexist.

However, measurement and identification perform satisfactorily only in the case of a single emission in the bandwidth of analysis. The detection of a situation of multi-emission is thus an important function of a system of spectrum monitoring and measurement.

##### 2.2 Recognition and processing of the situations of multi-emissions

To detect a situation of multi-emissions, it is recommended to use suitable representations of the signal. In a number of cases, a simple parametric visual representation makes it possible to diagnose the situations of multi-emissions, and to isolate the energy patterns corresponding to the various signals.

For that, one recommends to associate three types of analysis and visual representations illustrated hereafter:

- frequency analysis to segment the spectrum on an energy criterion;

- time analysis associated with the frequency analysis, for example, in order not to divide the spectrum of a multicarrier emission;
- the time/frequency analysis which makes it possible to recognize and to associate the various contributions of a burst signal or to recognize a signal variable in time and frequency.

For example, the time frequency representations make it possible to distinguish the case of a mono-source modulated frequency shift keying (FSK) signal from the case of continuous tone FSK signal (ex: OFDM signal), or the case of multiple narrow-band.

### 2.3 Separation of the signals – Separation of the energy patterns

One recommends to separate the signal components corresponding to a source using filtering functions in angular domain (use of a direction finder), and in the time, frequency and time/frequency domains (zoom functions, filtering with possibility either of choosing an existing filter or to define a filter, cut/paste/concatenate functions, etc. are advantageous features to this effect). These techniques are sufficient in the majority of the cases usually encountered in spectrum monitoring.

Figure 8 shows an example of the extraction of two burst signals by filtering in the time/frequency domain as well as the separation of one of the burst signal after time filtering.

## 3 Identification of an analogue signal

To identify an analogue signal, one recommends to associate four steps:

*Step 1:* direct listening to the demodulated AM, FM or PM, method also suitable for analogue FDM and TDM signals after successive demodulations and filterings;

*Step 2:* direct display of the demodulated analogue TV signal (TV tuner/demodulator cards that are available at reasonable cost for plug-in into common desktop computers can be used to this effect), method which applies to analogue FDM signals after successive demodulations and filterings;

*Step 3:* automatic method: identification of speech using the long-term characteristics of speech (in particular dynamics), by calculations and histogram representations of the variance and of the peak values of the signal;

*Step 4:* simultaneously with the preceding methods:

- panoramic representations and time frequency display of the signal to determine its nature (AM, SSB, FM, FDM, etc.);
- a basic consultation of database of the analogue signals of radiocommunication;
- measurement techniques previously recommended for the determination of the parameters of modulation:
  - carrier and subcarrier frequencies in the case of FDM protocols;
  - bandwidth of the signal;
  - modulation characteristics (amplitude depth, frequency and phase deviations), measured according to ITU-R Recommendations in force.

## 4 Recognition of OFDM unknown signals by other correlation techniques

*Autocorrelation:* The autocorrelation of the signal  $x$  is:

$$\Gamma_{x,x}(\tau) = \frac{1}{IT} \cdot \left[ \int_{\zeta=\tau}^{\zeta=\tau+IT} x(\tau + \zeta)x^*(\zeta)d\zeta \right]$$

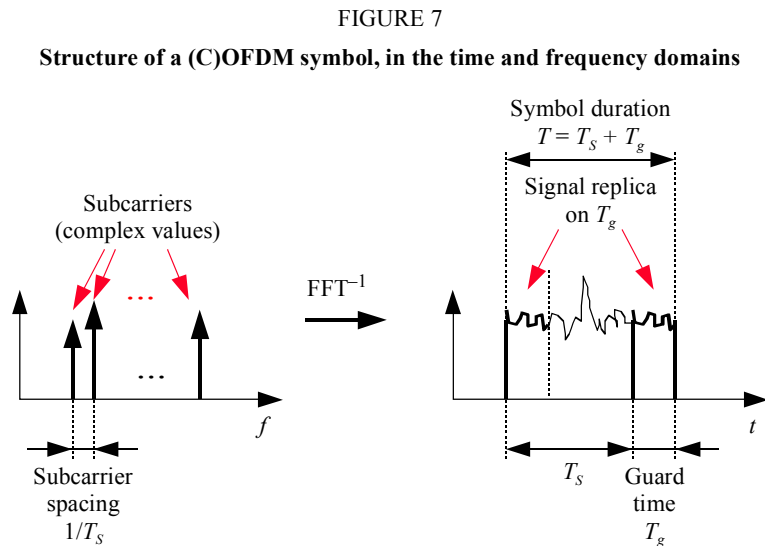
One application of the autocorrelation processing is the recognition of periodic sequences inside transmission signals that may be used as known signal in further processing. For example an accurate detection and recognition process that may be reached by combining autocorrelation calculation for recognition of a periodic known signal inside a CDMA transmission signal and by further intercorrelation processing based on this known signal.

*Cyclic autocorrelation:* The cyclic autocorrelation provides many advantages to face partially unknown signals as many OFDM and CDMA signals, and to point out cyclic properties of transmission signals of the signals. Its expression is:

$$\hat{\Gamma}_{x,x}(\alpha, \tau) = \frac{1}{IT} \cdot \left[ \int_{\zeta=\tau}^{\zeta=\tau+IT} x(\tau + \zeta)x^*(\zeta)e^{-2j\pi\alpha\zeta} \cdot d\zeta \right]$$

### 4.1 Application for the recognition of a (C)OFDM signal

The structure of symbols (C)OFDM is recalled in Fig. 7.



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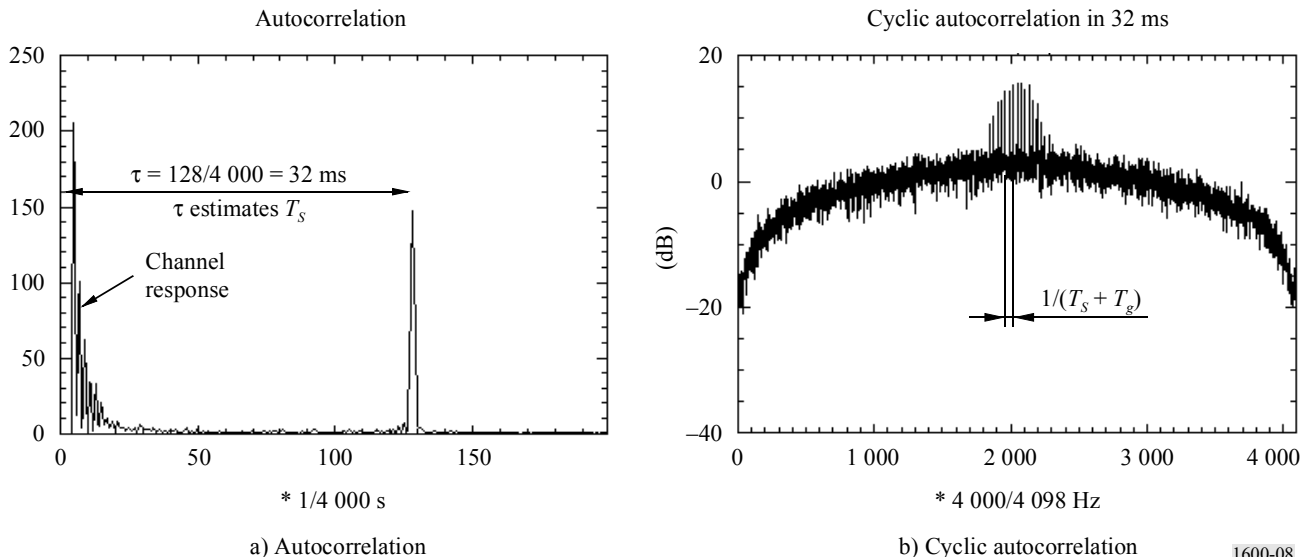
For the determination of the modulation rate and the symbol synchronization, it is possible to exploit the duplication of the beginning or the end of the symbol to constitute the part of the signal corresponding to the guard time. Thus, using the duplication of the signal allows application of efficient methods based on known signals. In the case of OFDM signal the basic mathematical function for calculation are the autocorrelation function and the cyclic autocorrelation function that were introduced before.

The practical implementation of these methods may be performed in three stages:

- Stage 1:* Counting of subcarriers, that can be made using a very fine spectral display (frequency resolution better than  $1/2 T_S$ ). One would recommend panoramic representations of the signal with variable spectral resolution (and integration time) and using a large number of points of calculation (for a good resolution and to interpolate the points of the spectrum by zero padding), associated with zoom functions and measurements by cursors.
- Stage 2:* Calculation of autocorrelation of the signal to reveal a peak corresponding to the delay  $\tau = T_S$  to determine spacing between subcarriers  $1/T_S$  (see Fig. 8). (It should be noted that the series of peaks corresponding to the echoes of the channel cannot be confused with the peak giving the symbol duration of the subcarriers because of their values.)
- Stage 3:* Calculation of cyclic autocorrelation for the delay  $\tau$  ( $\tau$  estimating  $T_S$ ) given by the autocorrelation so that correlated signal parts corresponding to the duplication of part of the symbol to constitute the guard time can be extracted (see Fig. 7):
- to confirm in addition the value of the symbol duration  $T_S$  (the cyclic autocorrelation calculated for a value of  $\tau$  other than  $T_S$  does not present characteristic peaks);
  - to determine the modulation speed of the subcarriers  $1/(T_S + T_g)$  and the guard time  $T_g$ .

FIGURE 8

## Correlation methods applied to a signal (C)OFDM



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The determination of the three parameters symbol duration,  $T_S$ , guard time  $T_g$ , and the number of subcarriers makes it possible in the majority of the cases to identify the (C)OFDM signal, and then carry out adequate measurements concerning the modulation of the subcarriers.

## 5 Identification of a digital signal other than (C)OFDM

To this effect, one recommends to proceed in three stages, very close to the process recommended to carry out measurements on *a priori* unknown broadband signals:

*Stage 1:* extraction of the carrier frequency and the modulation rate;

*Stage 2:* symbol (clock) synchronization and characterization of the modulation (standard, number of states, constellation of the linear modulations, etc.);

*Stage 3:* validation of the characteristics of the signal by demodulation, with the assistance of algorithms and associated diagrams of measurement on dedicated displays for:

- linear modulations on the one hand;
- non-linear modulations in addition.

### 5.1 Extraction of the carrier frequency and the symbol rate

According to the type of modulation, one recommends the methods summarized in Table 1.

TABLE 1

**Table recapitulating principal markers and methods to extract modulation characteristics of digital transmission signals**

Parameters to be estimated	Marker/process of measurement	Modulations concerned
Modulation – rate of asynchronous or synchronous modulation	Histogram of the duration of amplitude, $A_i$ , frequency, $F_i$ , and frequency, $F_i$ steps	On-off-keying (OOK), Unfiltered FSK and PSK
	Spectrum of amplitude, $A_i$	PSK filtered or not continuous phase modulation (CPM) or FSK after filtering
	Spectrum of frequency, $F_i$	Unfiltered FSK
	Zero crossing on frequency, $F_i$	FSK filtered or not
	Spectrum of signal module raised to power $N$ (2 or 4)	PSK and QAM filtered or not
	Spectrum of signal module raised to power $N$ after a severe filtering in frequency	FSK filtered or not
	Spectrum of the signal raised with power $N$ ( $N = 1/h$ , $h$ : modulation index)	CPM filtered or not
Carrier frequency Subcarrier frequencies	Spectral power density	Any modulation
	Histogram of frequency, $F_i$	FSK
	Average of frequency, $F_i$	FSK
	Spectrum raised to power $N$ (= 2 (PSK), 4 (QAM) or $1/h$ for CPM)	PSK, QAM, CPM
Bandwidth	Spectral power density	Any modulation
Frequency distance between subcarriers	Spectral power density	FSK, (C)OFDM
	Histogram of frequency, $F_i$	FSK, (C)OFDM

These methods must be associated with suitable representations of the signal after the various transforms it undergoes in order to extract the characteristics.

## 5.2 Characterization of the modulation of the signal

This characterization can effectively be carried out after the determination of the carrier frequency and the symbol rate of the signal using:

- a symbol synchronization display such as eye diagram and/or lattice of phase according to the type of modulation;
- a polar representation for the linear modulations (validation of synchronization, determination of the points of the constellation and the transitions between symbols);
- histogram representations for the phase or frequency modulations (validation of synchronization, determination of the number of sub-carriers).

According to the type of modulation, one recommends the methods summarized in Table 2.

TABLE 2

**Table recapitulating main markers methods to characterize the modulation of a digital signal**

Parameters to be estimated	Markers/process of measurement	Modulations concerned
Number of states	Polar display at decision time after equalization or matched filter	PSK, quaternary shift/phase shift keying (QPSK), $\pi/2$ DBPSK, $\pi/4$ DQPSK, QAM
	Histogram of $A_i$	OOK, amplitude shift keying (ASK), QAM
	Histogram of frequency, $F_i$	FSK
	Histogram of phase, $\Phi_i$	PSK
	Spectral power density	OFDM, (C)OFDM, multiplexing
Number of subcarriers or tones	Spectral power density	Any modulation
	Histogram of frequency, $F_i$	FSK
Frequency distance between subcarriers (shift for FSK)	Spectral power density	FSK, (C)OFDM
	Histogram of frequency, $F_i$	FSK, (C)OFDM
Symbol synchronization	Eye diagram $I/Q$ , $A_i/F_i$ + polar display	PSK and QAM filtered or not
	Eye diagram $A_i/F_i$ + histogram display frequency, $F_i$	FSK filtered or not
	Lattice of phase + histogram display frequency, $F_i$ and phase, $\Phi_i$	CPM filtered or not



### 5.3 Technical identification of a digital signal by association of measurement and demodulation methods

Once the extraction of the characteristics of the signal is carried out, it is recommended to validate this operation by a demodulation and an analysis of convergence.

For that, it is recommended to be able to have within the same tool a great number of digital demodulators, associated with various algorithms of channel equalization, and with charts allowing the analysis of the convergence of the process of demodulation.

- In the case of linear modulations, one recommends a representation of the demodulated symbol stream, corresponding error vector, points of the constellation (ideal points) and positions of the complex signal at the decision time on a polar representation, which allows an analysis of convergence, transitions between symbols, and possibly coding systems.
  - In the case of non-linear modulations, one recommends a histogram representation of the states of theoretical frequency or phase and states of frequency or phase of the signal to the decision time, for the analysis of the convergence of the demodulation process.
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