A MEASUREMENT SYSTEM FOR IN-SERVICE CHARACTERIZATION OF TELEPHONE-TYPE NETWORKS

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Abstract: A measurement system for in-service characterization of telephone-type networks is described. It is based on in-line Digital Signal Processing of the transmitted signal observed at an easily accessible test-point. The adopted techniques for the estimation of the considered parameters are summarized and discussed, together with an example of a measurement session performed on a real-life telephone-type network.

Keywords: quality, in-service non-intrusive measurement, telecommunication.

1 INTRODUCTION

In recent years the characterization of end-to-end voice transmission performances in telecommunication systems has become a primary issue both for public and for private network providers. In the assessment of the degree of quality of voice transmission provided by a telecommunication network the main aspect is represented by the customer opinion, that can be obtained by means of ad hoc inquiries among the users [1]. However, such inquiries should be supported by objective quality measurements frequently performed on the network.

In order to objectively assess transmission quality in a telephone-type network, a number of measurements should be performed. To this aim, many out-of-service tests have been proposed in the literature [2],[3],[4]. In out-of-service measurements a test signal is usually transmitted from one end of an idle network. Concurrently, the signal received at the other end is acquired and suitably processed in order to obtain significant information about the characteristics of the circuit under test.

Out-of-service tests usually allow determining the capability of the system to reconstruct a transmitted signal, for instance by providing the degree of distortion introduced by telephone channels. However, the cost due to frequently performed out-of-service measurements can be high, since the lines under test are not used by the customers. For this reason, the need for continuously monitoring the performances of communication networks has brought to the development of in-service measurement techniques [5].

By definition, in-service measurements are performed while the lines are actually in their operating mode. In this way, random or intermittent anomalies that usually are not detectable by means of out-of-service tests can be observed. Furthermore, the effective degree of quality of the service as observed by the users can be assessed.

Despite the interest in such measurements, a satisfactorily accurate instrument has not been proposed yet. In the paper, a system for in-service non-intrusive measurement is described. In the proposed system a Digital Signal Processor (DSP) based unit is connected to any easily-accessible test-point of the telecommunication system. It is provided with suitable interface circuits that allow the acquisition of telephone calls from the digital data stream of modern communication networks. Then, by means of numerical algorithms, the instrument simultaneously performs a number of measurements, including noise and signal parameter estimation, echo parameter extraction and interruption measurements on each observed telephone call.

The DSP-based device is provided with a front panel and hence can be used as a stand alone unit, or it can be used as the front-end of a more complex network supervision system. In this second case, it can exchange data with a remote host computer which collects the measurement results in a data base. Furthermore, the host unit allows the presentation to the user of a number of statistical indices related to
the data collected throughout the various measurements.

The outline of the paper is as follows. In Section 2 the system architecture is summarized, the parameters of interest for the characterization of the network are presented and the adopted estimation procedures are briefly described. In Section 3 experimental results that show the effectiveness of the proposed measurement system are reported and discussed.

2 IN-SERVICE CHARACTERIZATION OF A TELEPHONE CHANNEL

In order to obtain an in service characterization of the telephone channel under investigation a discrete-time version of both the transmitted and the received signal is obtained at an easily-accessible test point of the network. A simplified model of a telephone-type channel is reported in Fig. 1 [6]. According to international recommendations [5], in-service measurement devices can be connected only at test points in the four-wire trunk of a telecommunication system, such as the test point $P_1$ and $P_2$ in Fig.1. To this aim, the proposed system is provided with suitable interface circuits that allow the acquisition of the channels under test from the digital data stream of a communication network.

![Fig. 1 Simplified model of a telephone-type channel](image)

The samples acquired from the channels under test are processed by means of a Digital Signal Processor based unit, in order to evaluate the parameters of interest. Furthermore, the instrument can interact with a host computer by means of a modem through the telephone network, as sketched in Fig. 1. The measurement results collected by a number of devices spread all over the telephone network can then be stored in a data base in order to be post-processed by the user.

At the points $P_1$ and $P_2$ the following digital sequences referred to a sampling rate of $F_s=8$ KHz and quantization on 13 bit are observed:

$$x[n] = s_1[n] + w_1[n] + e_1[n], \quad y[n] = s_2[n] + w_2[n] + e_2[n]$$

(1)

where $s_1$, $s_2$ represent respectively the near-end talker signal and the far-end talker signal, $w_1$, $w_2$ account for the noise superimposed on the useful speech waveforms or for other impairments including clipping or interruptions while $e_1$, $e_2$ are echo signals. Then, $x[\cdot]$ and $y[\cdot]$ are processed in order to obtain meaningful parameters that describe the degree of quality of the network according to the techniques summarized in the following.

2.1 Noise parameters

A first set of meaningful parameters that provide a description of the overall noise superimposed on the
useful speech waveform is represented by the noise parameters. Noise present in telecommunication systems usually consists of a stationary component, well-described by a wide-sense stationary gaussian random process, and of a non-stationary component. According to [5], the stationary noise component can be well-characterized by providing the psfometric weighted noise level, i.e. the average root mean square of the weighted stationary noise. As concerns to the non-stationary noise component, a very important role is played by the so-called impulsive noise which can be well-summarized by few parameters, such as the arrival rate of impulses and the mean power of the impulsive component.

In order to estimate the noise parameters, the discrete-time signals $x[\cdot]$ and $y[\cdot]$ are segmented into active speech and noise intervals by using a suitably designed Bayesian classifier. Then, from the signal frames classified as noise it is estimated the Probability Density Function (PDF) of the Partial Noise Level, defined as the mean of $K$ successive squared observed samples. In fact, it can be shown that, if $K$ is properly chosen, the peak of the Partial Noise Level PDF is very close to the variance of the stationary noise component, also for arrival rates of the impulses as high as 50 impulses per second. Thus, a consistent estimate $s^2_s$ of the stationary noise variance $s^2_s$ can be easily evaluated by detecting the peak of the obtained Partial Noise Level PDF [7].

Successively, the hypothesis that the signal contained in an observed noise frame presents an impulse is tested, and is accepted or rejected with a predefined probability of error by adopting a hypothesis testing approach [7].

Finally, the frames containing impulsive noise are processed in order to estimate the arrival rate and the mean power of the impulses, while the stationary noise frames are used to determine a more accurate estimate of the psfometric weighted stationary noise level.

2.2 Voice parameters

Voice parameters describe the degree of quality associated to the useful information signal, i.e. speech. An important voice parameter is the Active Speech Level (ASL), defined as the mean square of the speech waveform. ASL can be obtained as the mean square value of the samples $x[\cdot]$ during speech intervals:

$$A\hat{S}L = \frac{1}{N_s} \sum_{s} x^2[i],$$

where $I_s$ is the union of all the frames classified as speech according to the quoted bayesian classifier used for noise measurement, while $N_s$ is the number of samples included in $I_s$. Furthermore, from the signal segmented into "Noise" and "Speech" intervals, other parameters of interest can be easily obtained, such as the Speech Activity Factor, i.e. the percentage of the total observation time in which active speech is present, or the Saturation Clipping, i.e. the percentage of samples in which the speech signal has saturated [5].

2.3 Echo parameters

Echo measurements assume a great interest both for the characterization of the network itself and for the tuning of the echo cancellers included in the network.

To the aim of echo measurements, a telecommunication system can be well-approximated by a linear system; thus, an echo path can be modeled as a linear system with impulse response $h[\cdot]$ usually well-represented by a pure delay and a non-zero signal with duration not longer than few ms. As a consequence, well-known linear system identification techniques can be adopted in order to evaluate the impulse response $h[\cdot]$, from which all the echo parameters of interest can be easily obtained [5].

The structure of the echo path impulse response can be advantageously exploited by the estimation procedure. To this aim, in the proposed instrument a two-stage procedure has been adopted. In the first
stage, a covariance analysis is performed on the incident speech signal $x[\cdot]$ due to the transmitter and the speech signal $y[\cdot]$ coming from the receiver. The presence of an echo is detected by comparing the peak of the covariance function $r_{xy}[\cdot]$ with a fixed threshold. The echo delay $t$ is estimated as the index that maximize the covariance function:

$$\hat{t} = \arg \max_k \rho_{xy}[k].$$

(3)

In the second stage, a delay line is introduced in order to compensate the echo delay. Then, an adaptive filtering technique is adopted in order to estimate the non-zero part of the echo path impulse response $h[\cdot]$. The delayed incident signal and its reflected version are the input signals to a FIR adaptive filter, which adjusts its coefficients in such a way to obtain a replica of the observed echo signal. When the filter has converged, its output is very close to the echo signal, while its coefficients represent an estimate of the non-zero part of the echo path impulse response. Since the duration of the non-zero part of the echo path is not longer than few ms, a low order FIR adaptive filter can be used. From the estimated echo path impulse response $h[\cdot]$, the parameters of interest are easily obtained [5].

2.4 Interruption measurements

An interruption is defined as a sudden change in level or attenuation of the transmitted signal for a short duration time-interval. Interruptions greatly affect the degree of quality provided to the users by a telecommunication system and assume a great interest especially when monitoring a mobile telecommunication system. Conversely, in the Public Switched Telephone Network, interruptions in the flow of the speech signals are seldom observed and are due to front-end clipping [5].

The duration of interruptions can be well-modeled as a random variable uniformly distributed in a pre-defined range, roughly equal to 50 ms - 1s. The time between two successive interruptions can be modeled as a geometric random variable, whose mean value represents the mean time between two interruption. Furthermore, during an interruption, the observed signal can be modeled as a white gaussian noise, whose level is lower than the level of the noise superimposed to the useful speech signal.

By adopting such model, interruptions can be completely characterized by providing the number of occurrences during a telephone call or a shorter observation period, and by the interruption duration and the signal power during an interruption.

In order to detect interruptions, the signal frames classified as noise are segmented in short analysis periods of equal duration. The noise level in each analysis frame is then compared with the estimated noise level obtained as described in sub-section 2.1. Finally, from each interruption the parameters of interest can be easily obtained.

3 EXPERIMENTAL RESULTS

Many experiments have been performed in order to verify the effectiveness of the proposed measurement system. In the experiments, speech signals recorded in a controlled environment have been considered, as well as computer-generated noise and real-life disturbances. For the sake of simplicity, in the following, only an example of a measurement session on a real-life data stream is reported.

In the considered example, 500 telephone calls obtained from a real-life communication link have been acquired and observed for 1 minute each. The Psophometric Noise Level, the Active Speech Level, and the echo parameters, i.e. the Echo Delay and the Echo Path Loss have been evaluated for each 1-minute telephone-call. Successively, the obtained measurements have been organized into histograms, in order to estimate the Probability Density Function of the evaluated parameters.
In Fig.2 (a) and Fig.2 (b) the histograms of the obtained Psometric Noise Level (PNL) and Active Speech Level (ASL) estimates, expressed in dBm [5], are respectively reported. It can be seen that the obtained PNL measurements are spread around a peak close to -60 dBm, while the ASL estimates are clustered around 14 - 16 dBm. In Fig. 2 (c) the histogram of the Echo Delay measurements (expressed in ms) is reported, while in Fig. 2 (d) the histogram of the corresponding Echo Path Loss measurements (expressed in dB) is shown. In the considered example, echo has been detected in about 50 observed telephone call out of the total. The obtained Echo Delays are all in the range 2-20 ms, while the corresponding Echo Path Loss are in the range -10 - -26 dB.

In the considered case, the level of the useful signal, i.e. the ASL has always been sufficiently high so as to provide a good intelligibility of the transmitted speech. Furthermore, the PNL has never reached high values. On the other hand, echo has been present in a high rate of the observed telephone calls, even though short echo delays have been observed. In some cases, the EPL has been high: this may be due to the absence of echo cancellers in the network under test or to their unsatisfactory behaviour.

4 CONCLUSIONS

In the paper, a measurement system for in-service characterization of telephone-type networks has been described. It is based on in-line Digital Signal Processing of the transmitted signal observed at an easily accessible test-point. The techniques for the estimation of the considered measurement parameters have been summarized and discussed together with a concise description of the system architecture. An example of a measurement session on a real-life telephone-type network has been reported and briefly
discussed.

REFERENCES


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