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Telephone Call Duration Measurement Reference System

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***Abstract*—The problems about telecommunication networks call duration measurement equipment validation and verification procedures are discussed in the paper. Because the maximum permissible error of network's call duration measurement system comprises many error terms that depend on telecommunication network structure, technology, call processing, etc., the "end–to-end" testing procedure have to be used. Therefore special reference system providing traceability of time measurements in telecommunications and measuring the duration of the test calls independent of network's equipment, utilizing "end-to-end" measurement principle, was developed. The structure and operation principles of the reference system are presented in details in the paper. Evaluated call duration measurement expanded uncertainty and systematic correction of the developed reference system is less than 0.1 s.**

***Index Terms*—Measurement techniques, metrology, telephone equipment, time measurement.**

I. INTRODUCTION

Time measurements in telecommunications fall within legal metrology scope because telecommunication companies still charge their consumers for the measured duration of the service, despite the fact that there is a variety of flexible service taxation used today.

From service taxing point of view telephone call is characterized by two parameters: starting time moment and duration. Starting time moment is a time-stamp taken immediately after the channel is established and available for communication between subscribers. The importance of starting time moment arises during the changes of tax rates. Today the telecommunication networks are synchronized with global positioning system (GPS) time or with the time supplied by national laboratories keeping the atomic time standards. Therefore the time used in the networks for call starting moments is traceable to the most accurate time sources. Actual call duration is another parameter of the service, meaning the period of time for which each "end-to-end" connection is available to both consumers for communication and is formed by these consumers entering and leaving their conversation, communicating from different, geographically distant, locations [1].

Call parameters measurements are made by telephone call duration measurement and billing systems (CMBS), which are integrated structural element of every telecommunication network. The functions of the CMBS are the following:

– Measurement of call duration;
– Call start time-stamp capture;
– Billing according to measurement results and call tariffs.

But the question arises – how accurately and reliably bills for the services are provided by the operators of telecommunication networks to the consumers? Every end-user is interested that the bills of telecommunication network operator would correspond to the actual duration of the service that has been used. The metrological control, which allows ensuring that the call service duration measurements, taken by the telecommunication network equipment, and charging corresponds to the actual amount of consumed services, is necessary.

During metrological procedures of telecommunication network CMBS the error of call duration measurement accordance with the maximum permissible error (MPE) is assessed. It is important to note, that the MPE of the network's measurement equipment is neither equal to resolution nor error of the timer of the particular station (server) of the network servicing the call and performing duration measurements. However, telecommunication companies very often report these latter parameters as main accuracy characteristics. In fact the MPE of network's call duration measurement system comprises many error terms that depend on telecommunication network structure, technology, call processing, etc. The error of the station timer, specified by the manufacturer is only one of them. But the problem is that the MPE of the particular CMBS is not known, since every telecommunication network has a unique structure, i.e. it have to be treated as a unique measuring instrument. Therefore one of the tasks for the approval (or control) body is the CMBS type examination and MPE determination during validation procedure [2].

II. METHODOLOGY

Keeping in mind the call duration error sources mentioned above the actual call duration can be biased by the network structural elements or operation principles and these changes cannot reasonably be attributed to the end-user. In order to investigate all network and internetwork links and their effect on the call duration measurement result, the "end-to-end" testing principle for CMBS metrological procedures was proposed [2]. In this case the whole telecommunication

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network is viewed as "black box" with two inputs/outputs – access points for subscribers A and B. This simplified model allows taking measurements of actual call duration, i.e. time period when communication channel available for both subscribers, measuring at calling subscriber place.

It is obvious that the validation or verification procedure of such complex system could not be done with ordinary calibrated time interval meter. Special reference system providing traceability of time measurements in telecommunications and measuring the duration of test calls independent of network operator's equipment utilizing proposed "end-to-end" call duration measurement principle is needed. The telephone call duration measurement reference system was developed in Metrology Institute, Kaunas University of Technology for validation and verification of telecommunication networks call duration measurement equipment. The structure of the system and operation principle is presented in details in the following sections.

III. Structure of the Reference System

Telecommunication networks are complex systems with a number of different technologies used for voice transmission, including public switched telephone network (PSTN) with analog and digital stations, voice over IP network (VoIP), global standard for mobile communications network (GSM), that are interlaced with each other and have different characteristics of signals, transmission, switching, signalling, etc. Therefore one of the requirements for reference system is invariance to the technology of telecommunication network.

Considering the said and to avoid cumbersomeness the reference system was designed and developed as two separate subsystems:

1. Subsystem for fixed telecommunication networks (PSTN analog, PSTN digital and VoIP);
2. Subsystem for mobile telecommunication networks (GSM).

Nevertheless from the work algorithm point of view these two subsystems are compatible, the only difference being the physical connectivity to different technology networks.

To embody the "end-to-end" testing principle and to establish a connection (call) between two subscribers, each subsystem comprises two different types of equipment:

1. Equipment that initiates the test calls and takes measurements; it is connected to the telecommunication network at subscriber A access point (equipment of subscriber A) (see Fig. 1);
2. Equipment that answers to the test calls; it is connected to the telecommunication network at subscriber B access point (equipment of subscriber B) (see Fig. 2).

The system structure is based on ARM Cortex M3 – central processing unit (CPU), which communicates with PC special software (subscriber A equipment only), telephone line module or GSM module, DTMF controller and runs the main algorithm. The timer interval meter is realized on quartz clocked CPU timer with highest interrupt priority, which was calibrated in the laboratory in order to ensure traceability in time measurements. The analog signal part of the system is controlled by Atmega 16 – analog signal processing unit (APU). The APU sets up the DTMF receiver, analyses analog audio signal (after signal conversion from analog to digital), detects lost connection (fixed telecommunication networks subsystem), and controls amplifier gain to ensure correct signal reception and detection.

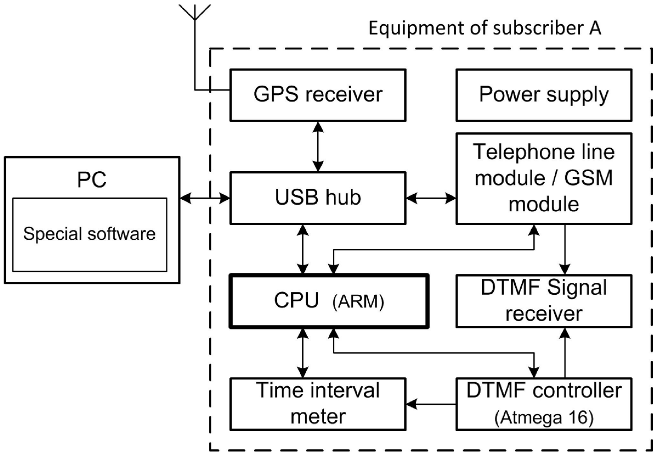

Fig. 1. The structure of subscriber A equipment (both subsystems).

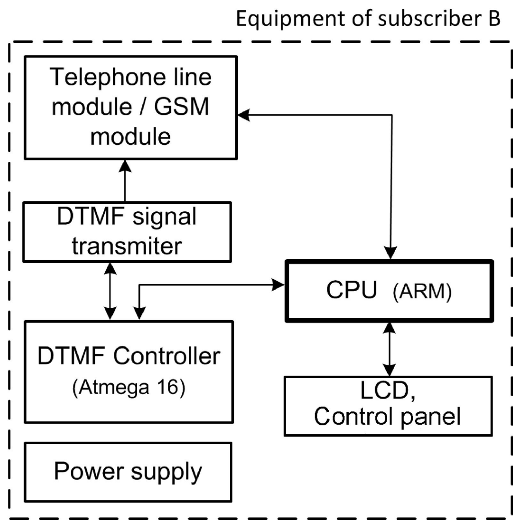

Fig. 2. The structure of subscriber A equipment (both subsystems).

The equipment of subscriber A of both subsystems has also integrated global positioning system (GPS) receiver (SiRF III) with active antenna for GPS time reception. The documented GPS synchronized time accuracy is 1 μs.

The functions of equipment of subscriber A (Fig. 1) are presented below:

– Full system control through graphical user interface (PC software);
– Fully automatic or manual test calls initiation modes;
– Call start time-stamp capture using GPS time;
– Call duration measurement;
– Analysis of connection channel signal quality and level;
– Detection of telecommunication station signalling;
– Storage of measurement results in data base format;
– Automatic formation of testing protocols;
– Events, processes and efficacy monitoring;
– Realization of calibration and other specific work modes.

Fig. 3. The developed telephone call duration measurement reference system.

The functions of equipment of subscriber B (Fig. 2) are the following:

– Implemented user interface using integrated LCD and control panel with a set of buttons for convenient equipment control;
– Automatic answering to incoming test calls;
– Generation and transmission of special signals;
– Random call duration formation from 2 s to 900 s, regarding given call duration distribution of particular operator telecommunication network and chosen work scenario;
– Stand-alone operation without inspector interference;
– Events and processes monitoring.

The developed reference system for validation and verification of telecommunication networks call duration measurement equipment is shown in the Fig. 3.

IV. Principle of Operation

The operation principle of the developed verification system is based on transmission of special signals (multi-sine – DTMF) through opened communication channel [3]. The call duration measurement lasts as long as the transmitted signal is detected. Below in this section the whole measurement process is presented in details.

Subscriber A and subscriber B pieces of equipment are connected to the network at chosen locations. Equipment of subscriber B acts as an “auto-answering generator”, which is designed to work autonomously and equipment of subscriber A is a “signal decoding time interval meter”, controlled by inspector.

The measurement process, consisting of four periods, is shown as the time diagram in Fig. 4. During first period of the process inspector starts initiating test call (either in automatic or manual mode), equipment of subscriber B is idle at that time. After setting up the test call, the telecommunication network begins connection procedure of these two subscribers (second period of the process). The equipment of subscriber B detects the incoming call, answers and immediately starts sending special signal through the opened channel (third period of the process starts). This signal is received and decoded by the equipment of subscriber A. The call duration measurement begins. As it could be seen from the time diagram there is a delay (“Delay 1” in Fig. 4) between real answer and signal detection which depends on the network characteristics.

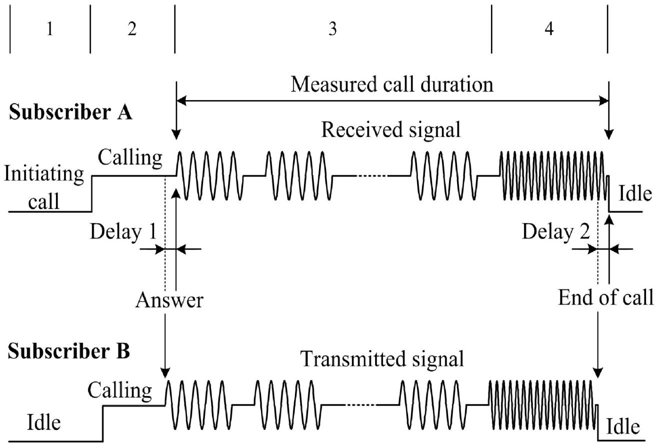

Fig. 4. The reference system operation time diagram.

There are several pre-programmed work scenarios of the equipment B: short fixed duration calls (4 s), middle duration calls (60 s), long duration calls (900 s), random duration calls, regarding the call duration distribution of the particular telecommunication network, etc. Subject to the chosen work scenario, the equipment of subscriber B calculates the duration of the call.

When the test call is coming to an end, the fourth period of call process starts: the signal generated by the equipment of subscriber B is changed for a short period of time (0,5 s) to inform the equipment of subscriber A about this coming moment. After that equipment B frees the communication channel and again the received signal is delayed (“Delay 2” in Fig. 4). These two delays can be different in general. The test call duration measurement is finished.

It should be mentioned that other testing modes (e.g. when subscriber A ends call) are also implemented.

Several aspects concerning the measurement process should be discussed:

1. The pulse-like signal transmitted by equipment of subscriber B is the result of the specificity of the fixed networks (PSTN, VoIP): the pause in pulsed signal serves for listening and detection of connection loss signal from the telecommunication station (or server) in order to hang up and free the channel.
2. There could be two types of errors:

a) The connection is cut and the signal is lost as a result;

b) The connection is not cut, but the signal is undecodable because of bad signal to noise ratio (or signal loss).

The results of such test calls duration measurements cannot be used as reference and therefore they should be marked as unsuitable and rejected. The earlier mentioned signal change (see fourth period of time diagram, Fig. 4) in the end of the test call is used for this purpose: if the signal is lost (undecodable) and there was not detected the signal change indicating the end of the test call, it is clear that the test call ended abnormally and therefore is rejected.

V. MEASUREMENT UNCERTAINTY

The expanded combined uncertainty and systematic effect correction of test call duration measurements were evaluated using the following expression [4]:

$$U = (\Delta_{cal} + \Delta_{dtmf} + \Delta_{alg}) \pm k_p u_c, \quad (1)$$

$$u_c = \sqrt{[u_e^2 + u_{cal}^2] + u_{alg}^2 + u_{osc}^2}, \quad (2)$$

where Δ_{cal} is the time interval meter correction (evaluated during calibration), u_{cal} is the time interval meter calibration standard uncertainty, u_e is the standard uncertainty of the reference equipment, used in calibration procedure, Δ_{dtmf} is the correction related to DTMF receiver signal detection and decoding time and DTMF receiver signal loss time difference, Δ_{alg} and u_{alg} are the correction and the standard uncertainty related to the work algorithm of main CPU respectively, u_{osc} is the standard uncertainty related to the resolution of the oscilloscope, k is the coefficient of uncertainty expansion, related to chosen coverage probability p.

The uncertainty terms and corrections are discussed shortly below.

The equipment of subscriber A of both subsystems was calibrated in the laboratory according to the calibration procedure. The absolute error results were distributed from 0 ms to -32 ms. Suppose, the probability density function is uniform, the systematic error is the centre of the function [5], i.e. is -16 ms and the correction Δ_{cal} is equal to 16 ms. The calculated standard uncertainty u_{cal} is 9.2 ms. The reference equipment uncertainty is very small and will be neglected.

The correction Δ_{dtmf} is related to DTMF signal reception circuitry could be written as:

$$\Delta_{dtmf} = \Delta_{start} - \Delta_{end}, \quad (3)$$

$$\begin{cases} \Delta_{start} = \Delta_{Sfix} + \Delta_{Sadj}, \\ \Delta_{end} = \Delta_{Efix} + \Delta_{Eadj}, \end{cases} \quad (4)$$

where Δ_{start} is the correction of the DTMF receiver related to reaction time of signal detection and decoding, Δ_{end} is the correction of the DTMF receiver related to reaction time of signal loss, Δ_{Sfix} and Δ_{Efix} are the fixed terms of Δ_{start} and Δ_{end} respectively, given by the DTMF chip manufacturer [6], Δ_{Sadj} and Δ_{Eadj} are the user adjustable terms of Δ_{start} and Δ_{end} respectively, which depend on chosen external capacitor (chosen capacitance value 100 nF) and resistor (chosen resistance value 47 kΩ). Calculated corrections Δ_{start} and Δ_{end} using (4) and formulae given in [6] for Δ_{Sadj} and Δ_{Eadj} estimation, are 17.3 ms and 11.8 ms respectively. According to the results, the correction Δ_{dtmf}, evaluated using (3), is equal to 5.5 ms.

The standard uncertainty related to the work algorithm of the equipment was evaluated using the measurement results taken with oscilloscope. The results of absolute error scattered between -10 ms and -50 ms. Therefore the systematic error is -30 ms (the correction Δ_{alg} is 30 ms). The calculated standard uncertainty u_{alg}, again assuming that the probability density function is uniform, is 11.5 ms. The standard uncertainty of oscilloscope resolution u_{osc} is 1.2 ms.

The expanded uncertainty of the developed reference system, choosing the coverage probability p equal to 95 % and therefore expansion factor k is equal to 2, is calculated using (1) and (2). The result is $U = (51.5 \pm 29.6)\ ms$.

VI. CONCLUSIONS

A unique reference system for telephone call duration measurement and call start time-stamp capture was developed for validation and verification procedures of call duration measurement and billing systems of various telecommunication networks (PSTN, VoIP, GSM).

Combined expanded uncertainty of the developed reference system call duration measurement is achieved less than 0.1 s.

The system design allows to make a large number of test calls in fully automatic mode using one of the pre-programed testing scenarios, save measurement results in the database, monitor system efficacy, control the test call process from start to end, analyse received signal quality, adapt the characteristics of the system to channel's signal level.

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