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Dear readers,

This issue of the Journal is devoted to communication technology and services. There are only few areas in the life which keep such rapid development and changes as the communication do. New challenges and new services come very quickly and push forward all communication processes.

The aim of the issue is to come close to selected problems of recent communication technologies. The content is divided into three parts.

First, some theoretical problems in optical communications are presented. Then we can switch to networking. On the Network architecture field there is unambiguous orientation on the IP platform. Although we can discuss if the IP is absolutely generous from the technical point of view there is no doubt that it is unparalleled technology from the implementation and service and technology integration point of view. Therefore a few contributions are dedicated to Next Generation Network and IP Multimedia Subsystem properties and solution. The last part is dedicated to the issues of protocols, services and their features.

I would like to thank all the authors who contributed to this issue of the Communications – Scientific Letters of the University of Zilina. I am also thankful to all the reviewers and editors.

Martin Vaculik

Eliska Jurisova – Jarmila Mullerova *

STUDY OF THE CHALCOGENIDE NONLINEAR FIBER BRAGG GRATINGS FOR ALL-OPTICAL SWITCHING

In this paper uniform and apodized nonlinear chalcogenide glass fiber Bragg gratings (FBG) as novel and promising devices for all-optical switching are investigated. The transmission response of these devices is discussed theoretically. The continuous wave in the investigated FBGs induces the nonlinear variation of the refractive index that causes changes in the FBG transmission characteristics. This effect can be modified depending on incident light power what is assumed to be due to the optical Kerr effect responsible for optical switching. The incident intensity thresholds of the bistable regimes were analyzed numerically to deduce possible switching properties of FBGs based on various chalcogenide glass samples. The influence of grating parameters on the optical bistability behaviour is researched. The nonlinear coupled mode theory was used for numerical investigation.

Keywords: *chalcogenide glass, fiber Bragg grating, optical bistability, optical switching.*

1. Introduction

Ultrafast optical switches are necessary parts of optical network nodes in high bit rate all-optical communication networks [1, 2] including systems of wavelength division multiplexing (WDM). Optical bistability (OB) in nonlinear media is a phenomenon that can be preferably used for all-optical switching. This optical phenomenon refers to the situation in which it is possible to have two stable output transmission states associated with a single input state, depending upon the history of the input. In nonlinear switches using the optical Kerr effect, the nonlinearity must be high and the nonlinear absorption must be low [1–3]. The switching behaviour is controlled by the intensity of the input optical pulse.

Chalcogenide glasses (ChGs) containing at least one of the atoms of Se, S or Te in their composition with the addition of As, Ge, Sb and Ga can form stable nonlinear media. Due to their large transparency in the infrared region, fibers fabricated from these glasses are valuable for transmission of high power IR light. ChGs possess high third-order Kerr nonlinearity up to 1000x higher than silica glass which makes them useful for nonlinear signal processing such as nonlinear switching in optical communication systems. ChGs are highly photosensitive with photo-induced refractive index change [4–6].

Fiber Bragg gratings (FBGs) are convenient structures for large refractive index change in ChGs. FBGs are primarily applied in optical communication networks as optical filters, WDM multiplexers/demultiplexers, optical circulators, optical add-drop multiplexers, all taking advantages of simple structure, small size, and low insertion loss. Besides optical switches nonlinear FBGs can be also used as distributed reflectors [7].

In this study, we numerically investigate the threshold of the bistable behaviour in nonlinear chalcogenide FBGs. We assume uniform and apodization profiles of the FBG refractive index. Characteristics of various samples taken from [4–6, 8] of ChGs containing different chalcogens are supposed to be the nonlinear media.

2. Theory

FBGs can be fabricated by the exposing photosensitive optical fiber with intensive ultraviolet light. It causes the variations of the refractive index along the FBG length L periodically (Fig. 1).

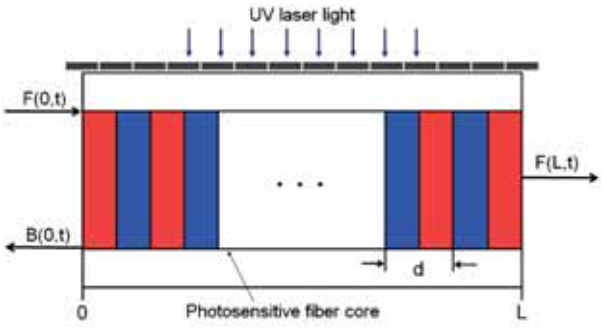

Fig. 1 Schematic diagram of FBG, where F and B represent envelopes of the forward and backward travelling waves respectively and d is the FBG period, L is the grating length

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The dependence of the refractive index on the position in the FBG (supposing the radiation is propagating in the z direction along the axis of the fiber) can be written as [9–11]:

$$n(z) = n_{eff} + V_n g(z) v\left(\cos\frac{2\pi z}{d}\right) + n_2\left|\vec{E}(z)\right|^2 \quad (1)$$

Here n_{eff} is the effective (average) refractive index, V_{n} is the depth of the modulation of the refractive index, $g(z)$ is an apodization function, v introduces the fringe visibility, n_2 is nonlinear refractive index and $\vec{E}(z)$ is the electric field.

Apodization is a well-known technique in which the refractive index is graded also in the direction perpendicular to the grating length. This technique can effectively remove the undesirable sidelobes in the reflection/transmission spectra of linear FBGs used as WDM multiplexers/demultiplexers or optical filters. The common apodization profiles are in Table 1 [11–12].

The light propagation in the nonlinear gratings is governed by the set of nonlinear coupled mode equations (NCMEs) [9–11, 13]:

$$\frac{\partial F}{\partial z} = i\left[\delta F + \kappa B + \gamma|F|^2 F + 2\gamma|B|^2 F\right] - \frac{c}{n_{eff}}\frac{\partial F}{\partial t} \quad (2a)$$

$$\frac{\partial B}{\partial z} = -i\left[\delta B + \kappa F + \gamma|B|^2 B + 2\gamma|F|^2 B\right] - \frac{c}{n_{eff}}\frac{\partial B}{\partial t} \quad (2b)$$

F and B are the amplitudes of the forward and backward modes, which determine the electric field inside the grating. $|F|^2$ and $|B|^2$ are proportional to the intensity of light. Coupling coefficient κ is defined as

$$\kappa = \frac{\pi V_n v g}{\lambda_{Bragg}} \quad (3)$$

where λ_{Bragg} is the Bragg (central) wavelength defined as $\lambda_{\text{Bragg}} = 2\pi_{\text{neff}} \cdot \delta$ represents the detuning between λ_{Bragg} and the carrier wavelength λ:

$$\delta = 2\pi n_{eff}\left(\frac{1}{\lambda} - \frac{1}{\lambda_{Bragg}}\right) \quad (4)$$

The parameter γ denotes the nonlinear Kerr effect coefficient:

$$\gamma = \frac{2\pi n_2}{\lambda_{Bragg}} \quad (5)$$

In linear regime $\gamma = 0$.

For the solving of NCMEs finite difference method (FDM) was used in order to obtain the relationship of the incident intensity and the transmittance of FBG. This method requires initial value points: $F(z = 0, t) = A(t)$, $F(z, 0) = 0$, $B(z, 0) = 0$. As initial conditions we assumed the zero amplitude of the backward wave: $B(z = L, t) = 0$.

OB occurs when the coupling coefficient, detuning parameter and nonlinearity coefficient match some conditions. This is significant to realize nonlinear switching in FBGs at the Bragg wavelength [4, 14]. The transmittance is defined as the ratio of the output and the incident light intensities.

3. Results and Discussion

In this section, we numerically simulate the bistable behaviour in nonlinear FBGs for different samples of ChGs the characteristics of which are depicted in Table 2. These experimental data are taken from [4–6, 8]. As it is supposed to apply nonlinear FBG switches mainly in the optical communication systems, it is important to set the investigation to the 3rd transmission spectral window (C band, around 1550 nm). Therefore we assume the propagating wave centred at the Bragg wavelength of 1550 nm. The investigated FBGs have parameters of the following values: $L = 10$ mm and $V_{\text{n}} = 0.0001$.

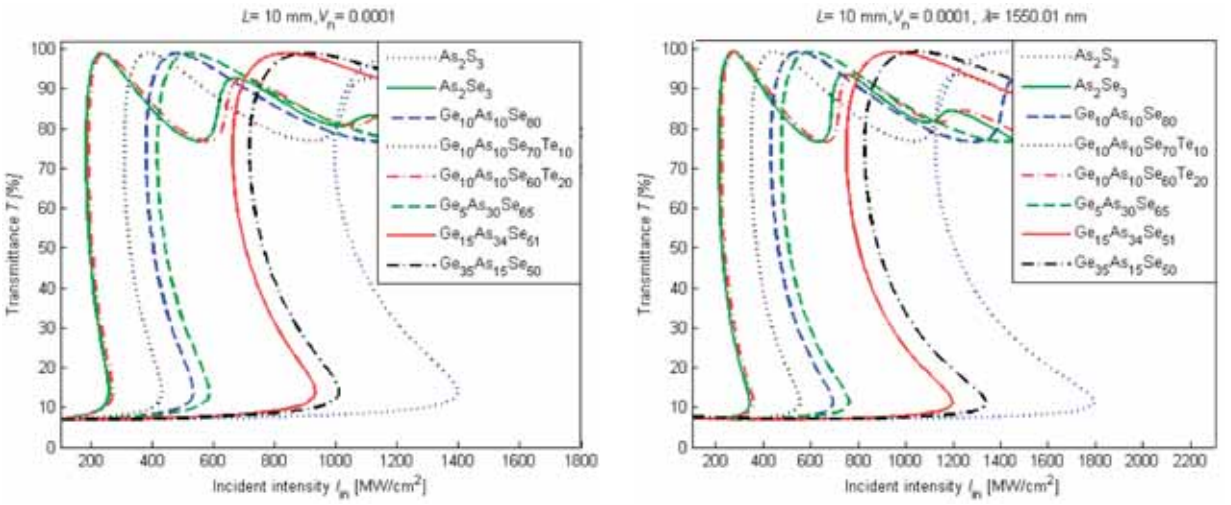
Parameters for selected ChGs samples taken from the literature [4–6, 8]. Table 2

Chalcogenide glass	n_{eff}	$n_2 \times 10^{-14}$ [cm^2/W]
As_2S_3	2.45	2.6
As_2Se_3	2.81	14
$Ge_{10}As_{10}Se_{80}$	2.58	6.8
$Ge_{10}As_{10}Se_{70}Te_{10}$	2.74	8.4
$Ge_{10}As_{10}Se_{60}Te_{20}$	2.90	13.4
$Ge_5As_{30}Se_{65}$	2.72	6.2
$Ge_{15}As_{34}Se_{51}$	2.64	3.9
$Ge_{35}As_{15}Se_{50}$	2.63	3.6

We suppose the stationary state of nonlinear systems. Then NCMEs (2) take the form

Three apodization profiles used for our investigations (G is the Gaussian parameter) Table 1

Gaussian profile	Sinc profile	Raised-cosine profile
$g(z) = \exp\left[G\left(\frac{z - \frac{L}{2}}{L}\right)^2\right]$	$g(z) = \frac{\sin(x)}{x}, x = \frac{2 \cdot \pi}{L}\left(z - \frac{L}{2}\right)$	$g(z) = \frac{1}{2}\left[1 + \cos\left(\frac{\pi}{L}\left(z - \frac{L}{2}\right)\right)\right]$



a) $\delta = 0$ *b) $\delta = -0.73489\ cm^{-1}$*

Fig. 2 The optical bistability curves for nonlinear FBGs based on ChGs. The values of their refractive and nonlinear index are in Table 2

$$\frac{\partial F}{\partial z} = i\left[\kappa B + \delta F + \gamma\left(|F|^2 + 2|B|^2\right)F\right] \tag{6a}$$

$$\frac{\partial B}{\partial z} = -i\left[\kappa F + \delta B + \gamma\left(|B|^2 + 2|F|^2\right)B\right] \tag{6b}$$

First, we present the results for FBGs supposed to be made from eight samples from Table 2 for the detuning parameter $\delta = 0$ (Fig. 2a)) and $\delta = -0.73489\ \mathrm{cm}^{-1}$ (Fig. 2b)). We see that all dependences manifest optical bistability (even multistability at higher input intensities). From Fig. 2 it is clear that the incident intensity required for the optical switching is minimal for As_2Se_3 and maximal for As_2S_3. Other suitable samples requiring moderate switching intensities seem to be ChGs with tellurium. As the samples differ by refractive indices, their important impact on the switching intensity can be deduced. As the nonlinear index increases, the incident intensity needed for switching decreases.

3.1 Results for the Chalcogenide glass As_2Se_3

In the next simulations we concentrate only on the sample of As_2Se_3 requiring the minimal switching intensity of all studied samples. In Fig. 3 the typical bistability curve only for this sample is depicted while the simulation parameters are the same as in Fig. 2a) and 2b).

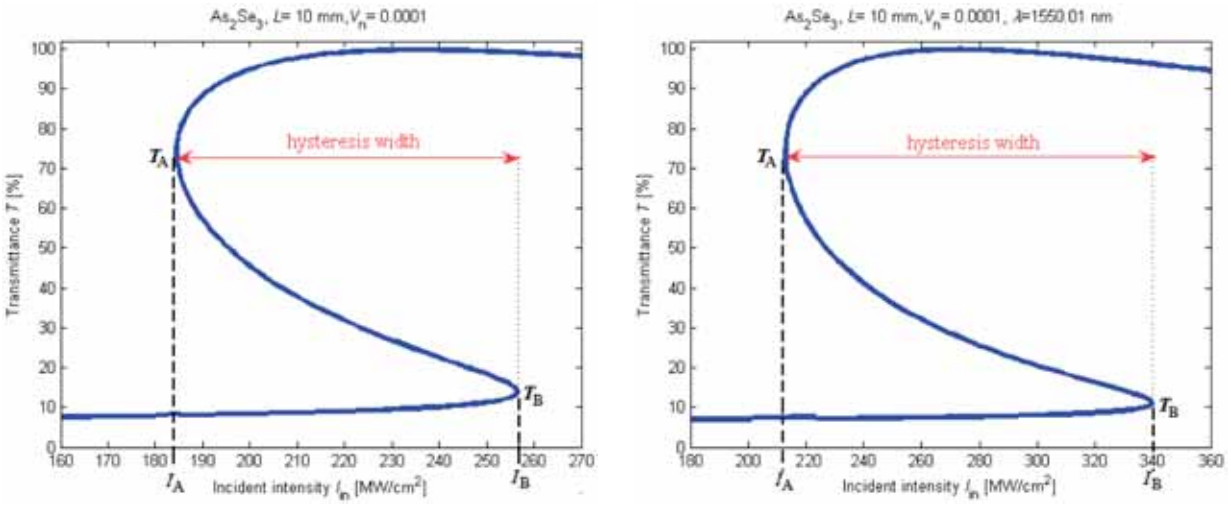
Fig. 3 represents the phenomenon of OB as the following change of the transmittance (T) as a result of the incident intensity (I_{in}). As I_{in} increases from zero, T follows the lower branch of the curve until Iin reaches the switching point (switch-on threshold), where the output makes a sudden jump to the upper branch. When I_{in} drops from high intensities, T can be brought back to the lower branch (switch-off threshold).The state of a linear increase of I_{in} is termed as the "OFF" state and the state of a sharp rise in Iin for $I_{in} > I_A$ is called the "ON" state. The difference of the transmission between "ON" and "OFF" states is the so-called on-off contrast. It indicates the transmittance interval ($T_A - T_B$) of the optical switching called the on-off contrast. The width of the S-shaped curve (the so-called hysteresis width, w) is defined as the difference between the switch-on (point I_B) and switch-off (point I_A) intensity thresholds.

It is clear that the switch-on intensity and the width of the hysteresis loop of OB increase with increasing the carrier wavelength (comparing Fig. 3a) and 3b)). The hysteresis width w for the case of $\delta = 0$ is $w \sim 72\ \mathrm{MW/cm}^2$ while for $\delta = -0.73489\mathrm{cm}^{-1}$ it is $w \sim 127\ \mathrm{MW/cm}^2$.

Next we present the numerical results for the influence of the Bragg wavelength on OB behaviour (Fig. 4). The carrier wavelength is suggested to be $\lambda = 1550.03$ nm. The increasing of this wavelength causes the increasing in the hysteresis width and in the switch-on intensity as well. Results in Fig. 4b) of Bragg wavelength variations are plotted for $\delta = -2.2046\ \mathrm{cm}^{-1}$. It is clear that in the all cases OB occurs. Therefore FBGs can be designed as optical switches for specified WDM channels.

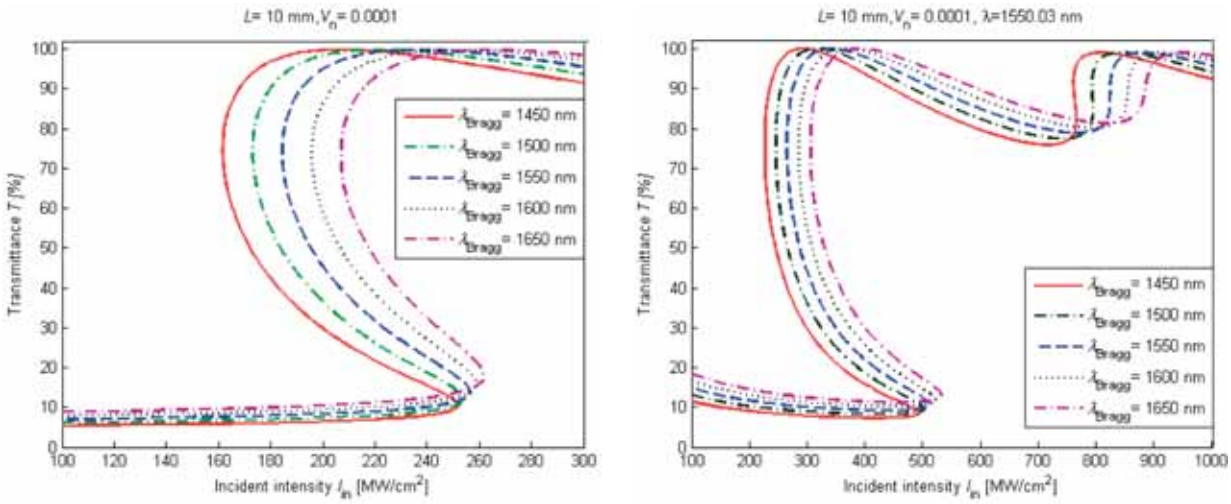
3.2 The Impact of the Apodization and Parameters of FBG

It is interesting to compare the nonlinear switching characteristics for uniform and apodized FBGs. The results for the uniform FBG and several apodized FBGs are presented in Fig. 5. The simulations show that the OB curve is present for all investigated index profiles, but the S-shape for the curves of OB, the values for



a) $\delta = 0$ *b) $\delta = -0.73489\ cm^{-1}$*

Fig. 3 The optical bistability curve for As_2Se_3



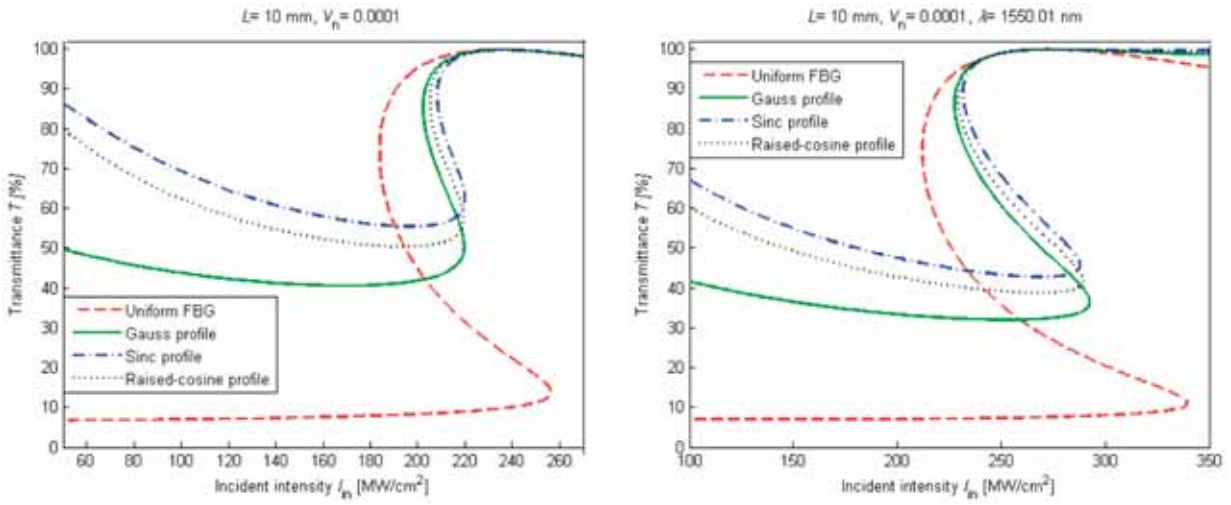
a) $\delta = 0$ *b) $\delta = -2.2046\ cm^{-1}$*

Fig. 4 The optical bistability curve for As_2Se_3 influenced by the change of λBragg

the transmittance, incident intensity and hysteresis width differ. The on-off contrast is higher for uniform gratings. We see that the apodization reduces the on-off contrast. Besides this we observe that apodized FBGs show the hysteresis width reduced in comparison with uniform FBGs. Both can be detrimental for optical switching that could fail or become accidental in these cases. Therefore we decided to alter some FBG parameters in wider intervals to study how they affect the switching conditions.

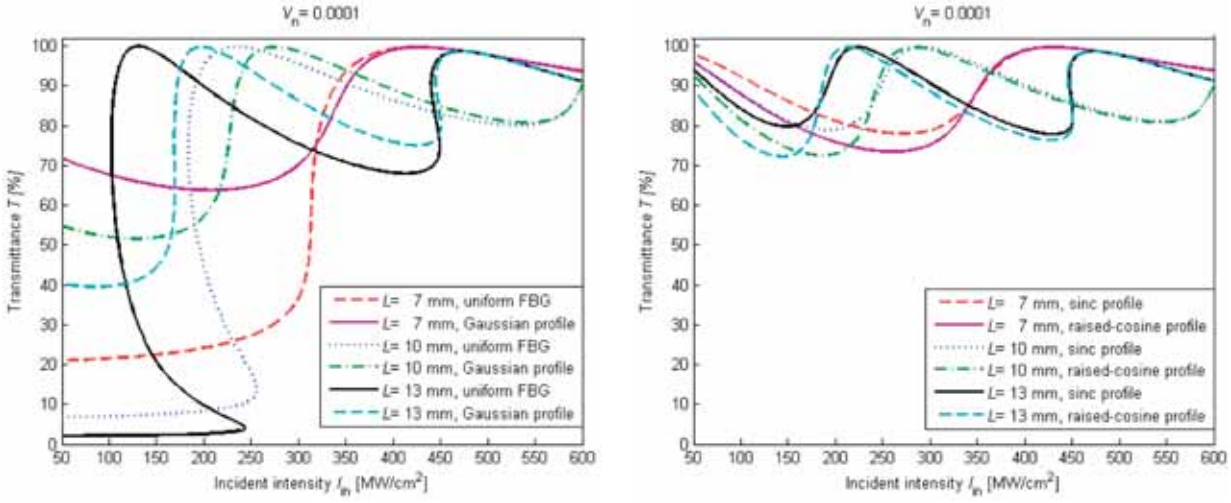
The investigation of the influence of the grating length L is depicted in Fig. 6. For the uniform FBG the length should be at least 7 mm to exhibit a slight bistability while for apodized FBGs of this length the bistability does not occur. Increasing on-off contrasts and hysteresis widths are observed with increasing L which is promising for optical switching. However, we found out that the on-off contrasts for sinc and raised-cosine apodization profiles are much smaller than those for Gaussian and uniform FBGs.

To improve the results we recommend optimizing the depth of the modulation of the refractive index V_n and/or the grating length L. Fig. 7 shows that higher V_n improves the on-off contrast



a) $\delta = 0$ *b) $\delta = -0.73489\ cm^{-1}$*

Fig. 5 Results comparison for As_2Se_3 for uniform FBG and apodization FBG



a) uniform and Gaussian apodized FBG *b) sinc and raised-cosine apodized FBG*

Fig. 6 Results comparison for As_2Se_3 of the uniform and apodized FBG by the change of L

of both uniform and Gaussian apodized FBGs. The changes are similar as those mapping the influence of the grating length.

For sinc and raised-cosine apodized FBGs, higher grating length is necessary to achieve desirable on-off contrasts and hysteresis widths. Therefore we simulate the influence of the depth of modulation at the grating length of $L = 20$ mm (Fig. 8) believing to achieve better results as in Fig. 6b). We see that with increasing V_n the on-off contrast is improved in comparison with data in Fig. 6b) although still not so pronounced as for uniform and Gaussian apodized FBGs.

4. Conclusion

In this paper we numerically demonstrated the optical bistability behaviour of the uniform and apodized chalcogenide nonlinear FBGs. The NCMEs were solved with FDM for the obtaining the switching threshold for our investigation. We found out the importance of FBG parameters as the FBG length, the depth of modulation of the refractive index and the Bragg wavelength, because they could have considerable influence on the shape of the hysteresis curve, the bistable threshold, the hysteresis width and the on-off

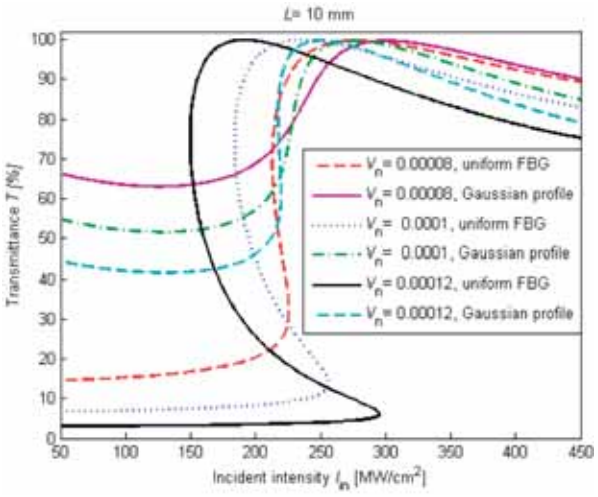


Fig. 7 OB in uniform and Gaussian apodized FBG, influence of varied V_n

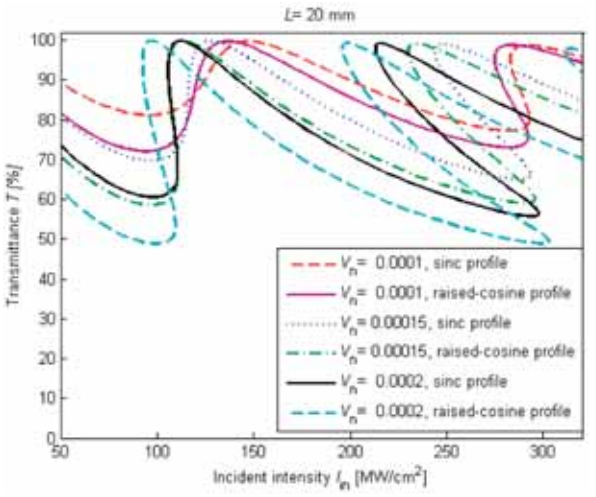


Fig. 8 OB in sinc- and raised-cosine apodized FBG, influence of varied V_n

contrast. Uniform FBGs appeared to be better for all-optical switching having higher on-off contrasts than the apodized FBGs.

Acknowledgement

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Juraj Smiesko – Jana Uramova *

ACCESS NODE DIMENSIONING FOR IPTV TRAFFIC USING EFFECTIVE BANDWIDTH

This article presents a dimensioning of capacity of access node (Black Diamond 12 804R) for IPTV source (6 channels of IPTV), which was one part of our project with Slovak Telecom. We describe IPTV traffic by two mathematic models and use Effective bandwidth for dimensioning of required link capacity with respect to QoS.

Keywords: *Markov Modulated Regular Process, M/D/1/queue, Network Transmission Line Dimensioning, Effective Bandwidth, Quality of Service, IPTV Traffic.*

1. Introduction

With the need of dimensioning the capacity of access node we met while working on a project for Slovak Telecom. It solves the problem of dimensioning link capacity in the network node. The challenge was to dimension the capacity for different scenarios, while we have available four different measurements of IPTV stream outgoing from the source of IPTV service Magio which was acquired in Slovak Telecom laboratory [1]. We had available measurements of monoscope and 1, 6 and 64 channels of IPTV. Our research shows that the recording of 6 channels of IPTV, we can approximate with Poisson distribution, while the coding creates a constant length packets. Proposal for capacity had to take into account the QoS parameters for the analyzed IPTV flow [2]. In agreement with the contracting authority of the research project, we considered only packet lost and delay. This work maps the procedure used in our search for suitable analytical method.

We deal with a model that assumes an infinite length buffer. Random variable q describes the steady-states distribution of the queue. Packet loss indicates the value $P_{lost} = P(q > n)$, where n is the recommended buffer size, which can be found. The excess value of X causes packets discard due to unacceptable delay. Input process $A(t)$ is Poisson process with intensity λ, $A(t) \sim Po(\lambda t)$, denotes the transmission link capacity c. We use a parameter $\rho = \lambda/c$, which represents the load or system utilization.

2. IPTV traffic description

The basic element of the network is traffic source. There are several concepts how to describe the packet flow generated by this traffic source. If we consider a discrete-time system, the traffic source will be described by arrival process $A(.)$ with increments $a(t)$, where $A(t)$ is the cumulative number of arrivals by the time t and $a(t)$ is the number of arrivals at time t for the arrival process $A(.)$ (increments at time t), $a(t) = A(t) - A(t-1)$. Let assume that there is no arrival at time $t-0, A(0) = a(0) = 0$ and packets from the traffic source are the same size. When we have the measurement of traffic which is sufficiently long, we can compute important statistical parameters, or create a corresponding stochastic model. Two basic parameters used for traffic description are average rate λ_{avg} and peak rate λ_{peak}. It is easy to compute these parameters from the measured traffic with length

$$\lambda_{avg} = \rho = \frac{1}{n}\sum_{i=1}^{n} a(i) = \frac{A(n)}{n}, \lambda_{peak} = \max_{0 \le t \le n}\left[a(t)\right]$$

We want to design a stochastic model for the arrival process description. We considered a time slot of 20 ms. The record with increments of packet stream with 6 channels of IPTV service Magio 2000 ms long is shown in Fig. 1. The entire record we worked with is 2 minutes long.

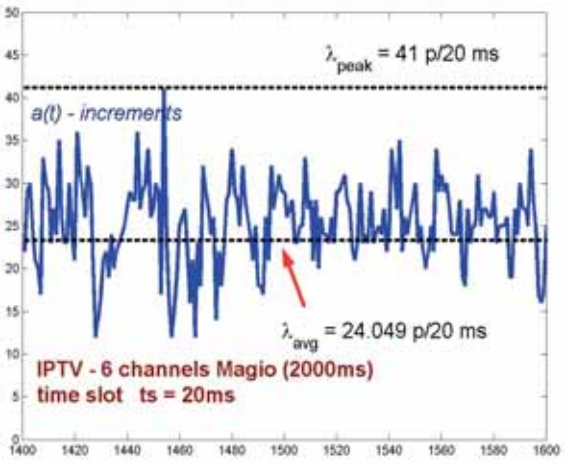

Fig. 1 6 channels of IPTV service Magio, time slot 20 ms, duration 2000 ms

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Our task is to find a suitable capacity c for the IPTV stream while we have to meet Quality of service, packet lost P_{lost} and maximum delay d. The value of capacity will be between the mean intensity and the peak value:

$$\lambda_{avg} \leq c \leq \lambda_{peak}$$

First we find a suitable stochastic model for the measured flow then we will deal with analytical methods for determining the capacity.

For the time slots, we calculated the empirical probability distribution of increments $a(t)$ and we compared their values with Bernoulli and Poisson models. In our research we did experiments with different lengths of time slots. We found that for the size of time slot less than 10 ms Bernoulli model is very close to the measured data, in all other cases, however, better approximation is achieved by Poisson model. Figure 2 illustrates the already mentioned case with a length of time slot of 20 ms.

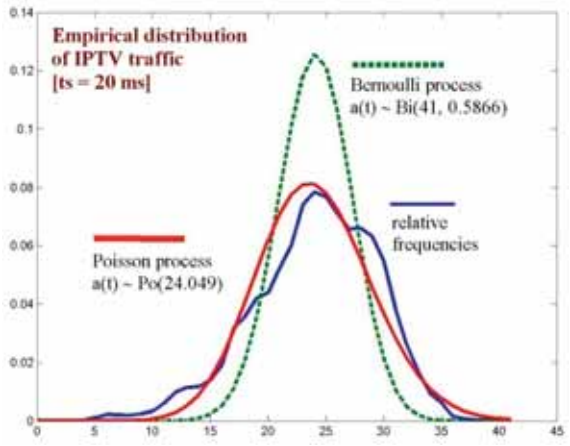

Fig. 2 Two stochastic model for IPTV traffic

For further analysis we decided to use a Poisson model for several reasons. In addition to the above distribution we took into account the fact that the Bernoulli model versus Poisson model provides an upper estimate for the probability of higher values in the measured flow. Our task was to allocate the capacity of the flow with respect to QoS, so we decided for a model with higher probability of peak traffic. Another crucial fact for the choice of the Poisson model was that we wanted to be able to use M/D/1 queueing model for the capacity dimensioning.

3. Queueing system M/D/1/queue

Queueing system M/D/1/queue is well known in Queueing Theory This system has a single queue, Poisson input process (M), constant service distribution (D). These parameters correspond to Poisson model of IPTV flow and constant packet length.

Let probabilities

$\pi_k = P(q = k)$ represent a steady-state distribution of a stabilized system. Parameter $\rho = \lambda/c$ denotes a system load. For the stabilized system $\rho < 1$ must be satisfied. Probabilities are derived recurrent equations, [3]:

$$\pi_0 = e^{-\rho}\,[\pi_0 + \pi_0],\quad k = 1, 2, 3, ...,$$

$$\pi_k = e^{-\rho}\left[[\pi_0 + \pi_1]\frac{\rho^k}{k!} + \sum_{i=1}^{k}\pi_{i+1}\frac{\rho^{k-i}}{(k-i)!}\right]$$

The formula for probability of packet loss $P_{lost} = P(q > n)$ for value $n = 3, 4, ...$ is:

$$p_{lost} = 1 - \left[e^{2\rho} - \rho e^{\rho}\right]\pi_0 - \\ - \sum_{k=3}^{n}\left[e^{\rho}\pi_{k-1} - \frac{\rho^{k-1}}{(k-1)!}e^{\rho}\pi_0 - \sum_{i=1}^{k-2}\pi_{i+1}\frac{\rho^{k-1-i}}{(k-1-i)!}\right] \quad (1)$$

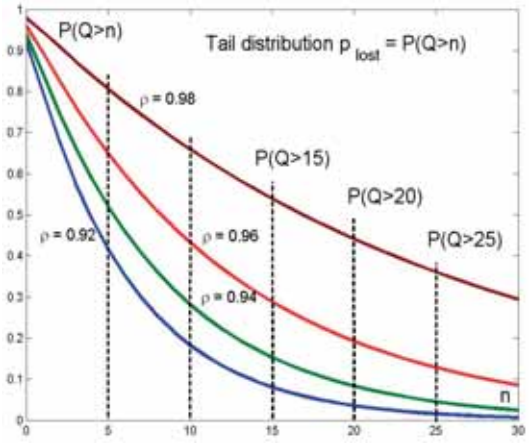

Fig. 3 Probability distribution and tail distribution of different values ρ

For the given values of ρ and P_{lost} we know how to design a maximum accepting buffer size n and consequently a maximum delay using probability queue distribution of model M/D/1. See Fig. 3 for illustration.

For example, let $\rho = 0.95$ and $P_{lost} = 0.001$. From the expression (1) we gain the maximum accepting queue size $n = 47$. Strictly speaking 47 is the first value when the tail distribution decreases under 0.001 $0.000961 < 0.001$. It's easy to compute the needed link capacity and maximum delay for the given average rate $\lambda = 1201.7(p/s)$: $c = \lambda/\rho \cong 1271$ p/s, $d = n/c = 37$ ms.

Our problem is the opposite. For the given average rate, lost probability and maximum delay (or acceptable buffer size) we have to compute a link capacity. But this means that the load of the link ρ is unknown and the calculation according to equation (1) is nec-

essary to use complicated iterative algorithms. For this reason we decided to use the Theory of Large Deviation Principles and Effective Bandwidth for a traffic flow description.

4. Effective Bandwidth

The concept of effective bandwidth (EB) has gained much attention due to the looming gain for network analysis and design. The effective bandwidth of a general cumulative arrival process has been defined [4] as

$$\alpha(\theta,t) = \frac{1}{\theta t}\sup_{s\geq 0} E\left[e^{\theta(A(s+t)-A(s))}\right] \qquad (2)$$

$0 < \theta, t < \infty$

depending upon the space parameter θ and the time parameter t. The effective bandwidth provides a convenient tool for analysis and description of data flows. Theory of Large Deviation Principles provides the tool for link dimension with respect to probability of packet loss (see [5])

$\alpha(\theta, t) = c \;\Leftrightarrow\; P(q > n) \asymp e^{\theta n}$ or

$$\lim_{n\to\infty}\frac{\ln P(q > r)}{n} = -\theta \qquad (3)$$

where constant n is size of queue (buffer) and variable q is steady-state length of queue. The relation (3) means that if we design a capacity of link equal to the effective bandwidth, than probability of buffer overflow decays exponentially with constant θ.

If the arrival process A(t) has a ***stationary increments*** a(i), we gain a simple form of the effective bandwidth:

$$\alpha(\theta,t) = \frac{1}{\theta t}\sup_{s\geq 0}\ln E\left[e^{\theta(A(s+t)-A(s))}\right] = \frac{1}{\theta t}\ln E\left[e^{\theta A(t)}\right] \qquad (4)$$

The effective bandwidth has very interesting properties. For example, the average rate, asymptotic variance, peak rate and burst period are included in its Taylor expansion [5]. Further, the value of EB in $\theta = \mathbf{0}$ equals to the average rate:

$$\alpha(0,t) = \frac{E\left[A(t)\right]}{t} = \lambda_{avg}$$

If the process A(t) has a ***stationary and above bounded increments,*** $\forall t$, $a(t) \leq \lambda_{peak}$, it is easy to show (using Jensen inequality [6]), that the effective bandwidth is between average rate and peak rate:

$$\lambda_{avg} \leq \alpha(\theta, t) \leq \lambda_{peak} \qquad (5)$$

Let process A(t) have ***stationary (identical) and independent increments*** (i.i.d) a(t). This means that the random variables a(t) have the same probability distribution and are mutually independent. Let $\varphi_a(\theta) = E[e^{\theta\alpha(t)}]$ be the moment generation function and $\lambda_a(\theta) = \ln \square\, \varphi_a(\theta)$ be the cumulative generation function of increments. The effective bandwidth has form:

$$\alpha(\theta,t) = \frac{1}{\theta t}\ln E\left[e^{\theta\sum_{t=1}^{t} i^x}\right] = \frac{1}{\theta t}\ln\varphi_a(\theta)^t = \frac{\lambda_a(\theta)}{\theta} \qquad (6)$$

The effective bandwidth of arrival process with i.i.d increments doesn't depend on time parameter and equals to scale cumulative generation function. Then we assign the effective bandwidth as α(θ). The formula for link dimensioning is

$$\alpha(\theta) = \frac{\lambda_a(\theta)}{\theta} = c \;\Leftrightarrow\; P(q > n) \asymp e^{-\theta n} \qquad (7)$$

In many cases of buffer dimensioning for real scenarios, the buffer size n is unknown but allowed delay for traffic is given. When we assume FIFO service policy [7] and put $c\theta = \lambda(\theta)$ we can redesign dimension formula (8) as follows:

$$d = \frac{n}{c} \;\Rightarrow\; P_{lost}(q > n) = e^{-\theta n} = e^{-\theta dc} = e^{-d\lambda_a(\theta)}$$

We replaced asymptotic decay "$\asymp$" by equality "=". It means we will obtain upper estimation of a queueing tail distribution. Cumulative generation function is strictly convex. This guarantees the existence of inverse function $\lambda^{-1}(.)$. We obtain the dependence between space parameter θ and QoS parameters:

$$\theta_0 = \lambda^{-1}\left(\frac{\ln p_{lost}}{-d}\right) \qquad (8)$$

Now we can design a link capacity using θ_0 : $c = \dfrac{\lambda(\theta_0)}{\theta_0}$

and we can guarantee upper estimate for values of maximum delay and packet loss probability. Equation (9) we call ***Formula for Link Dimensioning using Effective Bandwidth with respect to QoS.***

5. Dimensioning capacity for IPTV traffic

5.1. Poisson model

The main basic model for IP traffic is ***Poisson process***. This is process with i.i.d. increments with Poisson distribution. The effective bandwidth for Poisson process is:

$$\alpha(\theta) = \lambda\frac{e^{\theta} - 1}{\theta} \qquad (9)$$

For comparison with steady-state analysis of model M/D/1 we use same inputs parameters, $\rho = 0.95$ and $p_{lost} = 0.001$. Now we have to find a numeric solution for equation

$$\frac{1}{\rho} = \frac{y^{\theta} - 1}{\theta} \;\Rightarrow\; \theta = 0.1016$$

We compute an acceptable queue length and maximum delay:

$$p_{lost} = e^{-n\theta} \Rightarrow n = \frac{\ln p_{lost}}{-\theta} \doteq 68 \Rightarrow$$
$$d = \frac{n}{c} = \frac{68}{1265} = 54ms$$

Using Effective Bandwidth we derive the maximum delay 54 ms which is more than we have gained from the model using equation (1), 37 ms. On the other hand we can say that for a given setting we guarantee that the maximum delay no more than 54 ms.

Our goal was the opposite, we should determine the required link capacity for the QoS parameters. First, we can compute parameter θ for a given delay and packet loss using formula (9):

$$\theta_0 = \ln[d\lambda - \ln p_{lost}] - \ln[d\lambda] \quad (10)$$

Using the equation (8) we derive the explicit formula for link capacity:

$$c = F(d \cdot p_{lost}) = \frac{\ln p_{lost}}{d\left[\ln(d\lambda) - \ln\left[d\lambda - \ln p_{lost}\right]\right]} \quad (11)$$

Using the average rate of the traffic $\lambda = 1201.7(p/s)$ and specified QoS parameters $d = 100$ ms and $p_{lost} = 0.00001$ we get:

$$\theta_0 = 0.091,\ c = 1264.2\frac{p}{s},\ n = d \cdot c \doteq 127,\ \rho = 0.955$$

We get upper estimate of the required link capacity 1264.2 p/s, or load $\rho = 0.955$. We compare the result with the values obtained directly from the model M/D/1. When the load is 0.955 we observe with expression (1) when its value falls below p_{lost}, $P(q > n_0) \leq$ ≤ 0.00001.

We obtained the value of buffer size $n_0 = 126$. See Fig. 4.

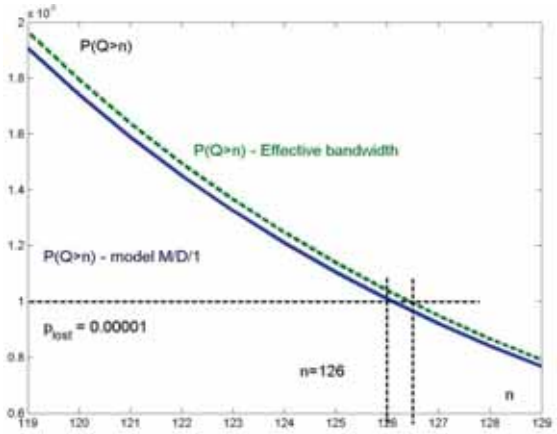

Fig. 4 Tail distribution for n = 119, ..., 129

This is a very pleasing result achieved using the Effective Bandwidth. We get the same result as we obtained by iterative algorithms when using the queue distribution of the model M/D/1. Using Effective Bandwidth only as the substitution into the explicit function of capacity was sufficient.

To get the most accurate results for estimating the capacity we will use a more detailed model of the process and describe the IP traffic flow using Markov Modulated Regular Process, MMRP.

5.2. Markov Modulated Regular Process

Markov modulated process (MMP) consists of the period On and period Off. If the process is in period On, it produces IP traffic, and if it is in the period Off, it means that nothing is transmitted. The simplest situation is when the process produces a regular (deterministic) flow of ones in period On (Fig. 5). Let's call this process as Markov Modulated Regular Process (MMRP). Whereas the switching between the states On and Off is controlled by Markov chain, MMRP is process with i.i.d. increments $a(i)$, which takes values 0 and 1 ([8]).

First state of the chain describes the period On, second state describes the period Off. Let's designate probabilities of transitions between the states as $p_{1.2} = \alpha$ and $p_{2.1} = \beta$. Let $\pi = (\pi_1, \pi_2)$ be a steady-state distribution of Markov chain. Using Queueing theory we compute values of steady-state probabilities ([2]):

Let's create a matrix of the intensities of transitions Q for Markov chain and equations for probability of states for a stabilized chain π:

$$P = \begin{pmatrix} 1-\alpha & \alpha \\ \beta & 1-\beta \end{pmatrix} \cdot \pi \cdot P = \pi \Rightarrow$$
$$\pi_1 = \frac{\beta}{\alpha + \beta}, \pi_2 = \frac{\alpha}{\alpha + \beta}$$

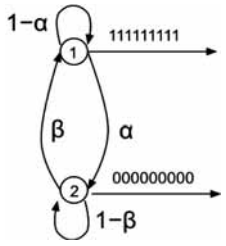

Fig. 5 Markov Modulated Regular Process

Let's calculate a probability distribution of increments in an elementary time slot:

$$p_1 = \pi_1(1 - \alpha) + \pi_2\beta = \pi_1,$$
$$p_0 = \pi_1\alpha + \pi_2(1 - \beta) = \pi_1,$$

Determine the probability distribution of increments in four slots. For slots longer that $m \geq 4$, the situation is even more complicated:

$$p_0 = \pi_2(1-\beta)^3$$

$$p_1 = (\pi_1\alpha + \pi_2\beta)(1-\beta)^2 + 2\alpha\beta(1-\beta)\pi_2$$

$$p_2 = (\pi_1\alpha + \pi_2\beta)(1-\alpha)(1-\beta) + \alpha\beta(\pi_1\alpha + \pi_2\beta + \\ + \pi_1(1-\beta) + \pi_2(1-\alpha))$$

$$p_3 = (\pi_1\alpha + \pi_2\beta)(1-\alpha)^2 + 2\alpha\beta(1-\alpha)\pi_1$$

$$p_4 = \pi_1(1-\alpha)^3$$

Our measured IPTV packet flow is sampled to such a maximum time slot so that we obtain no more than 4 packets in one slot. The searched size of time slot is 0.5 ms. The average rate is $\lambda_{avg} = 0.6(p/0.5\text{ms})$ and, of course, the peak rate is $\lambda_{peak} = 4(p/0.5\text{ms})$. The parameters of MMRP model are gained using Ordinary Least Squares – OLS:

$$\alpha = 0.8065, \beta = 0.1423 \Rightarrow \pi_1 = 0.14998, \pi_2 = 0.85002$$

We compare the obtained empirical probability distribution with the distribution of Poisson process and MMRP. The differential vector size between the empirical distribution and model distribution is used as the error of approximation:

Distribution	0	1	2	3	4	Error
Empirical	0.536	0.345	0.101	0.015	0.002	0
Poison	0.548	0.329	0.099	0.020	0.003	0.020964
MMRP	0.536	0.345	0.102	0.016	0.001	0.001370

When using the time slot size of 0.5 ms we obtained the absolute correlation between the distribution of MMRP model and measured data of 6 channels of IPTV service. We want to show suitable use of MMRP, thus we present a comparison of Effective bandwidths for all three processes. Effective bandwidth for 2-state MMRP has the form ([3]):

$$\alpha(\theta) = = \frac{1}{\theta}\ln\left[\frac{e^{\theta}(1+\alpha) + (1-\beta) + \sqrt{\left[e^{\theta}(1-\alpha) + (1-\beta)\right]^2 - 4e^{\theta}(1-\alpha-\beta)}}{2}\right] \quad (12)$$

Effective bandwidth estimate obtained from the measurements using estimates $\hat{p}_k$:

$$\alpha(\theta) = \frac{1}{\theta}\ln\left[\sum_{k=0}^{A} e^{\theta k}\hat{p}_k\right] \quad (13)$$

The Effective Bandwidth of MMRP model significantly better approximates the estimate of Effective Bandwidth of measured IPTV traffic (Fig. 6). Although the Effective Bandwidth of MMRP acquires lower values than the real Effective Bandwidth for higher values of scale parameter θ, but in real values of QoS parameters p_{lost} and d value of the scale parameter θ does not exceed the value $\theta_0 = 1$ (its our own empirical opinion).

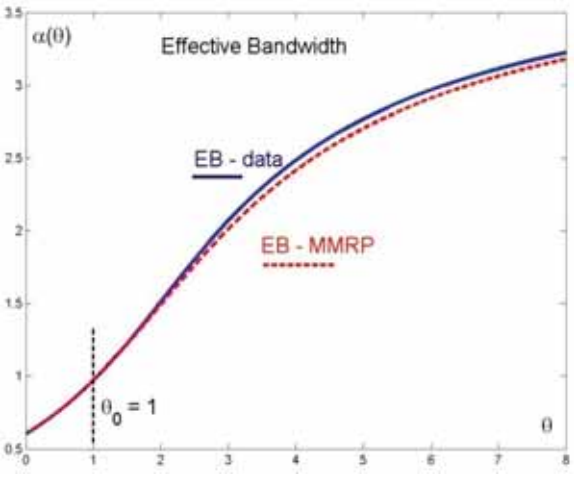

Fig. 6 Effective Bandwidth for MMRP and IPTV traffic

Formula (9) for MRRP model for determining the scale parameter θ has the form:

$$\theta_0 = \ln\left[p_{lost}^{-1/d}\right] + \ln\left[p_{lost}^{-1/d} - 1 + \beta\right] - \\ - \ln\left[p_{lost}^{-1/d}(1-\alpha) - 1 + \alpha + \beta\right]$$

For specified parameters $p_{lost} = 0.00001$ a $d = 100$ ms we get:

$$\theta_0 = 0.32819,\ c = \alpha(\theta_0) = 1403.2\frac{p}{s},\ n = d \cdot c \doteq 140,$$

$$\rho = 0.856$$

The recommended link capacity with respect to specified QoS parameters is substantially higher for MRRP than what we obtained using the Poisson process. However, we can consider this value $c = 1403.2$(p/s) as more relevant, thanks to the equality of MMRP model with distribution of the measured data of IPTV traffic.

6. Conclusion

Our task was to recommend the capacity for access nodes to the core network for measured IPTV traffic flows with respect to QoS parameters. This task was related to research for Slovak Telecom. In this article we analyzed 6 channels of IPTV service for which we came to a Poisson process as an ideal model. But the dimensioning of access node using the classical model M/D/1 is proved impractical (frame sizes of IPTV traffic were the same through the use of coding). Therefore, we decided to use the Effective Bandwidth for access node dimensioning. For our purposes we derived the Formula for Link Dimensioning with respect to QoS. To get

better results we used the Markov Modulated Regular Process (in addition to Poisson process) which reached the absolute conformity with the measured data. The obtained results were tested by simulations and recommended to the contracting authority of the research project as relevant. For the theoretical contribution of our work we consider the derivation of formulas for calculation of a scale parameter using the inverse cumulative function, providing effective bandwidth for the traffic. We found that for specific parameters typical for IPTV (it means very low probability of packet loss and a relatively high maximum delay) an estimate of the capacity obtained through the Effective bandwidth is very close to the value obtained from the queueing model. In parallel, we analyze the IPTV service with a different number of channels, in addition to 6 channels. The results of this research will be included in subsequent articles.

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Jan Holub – Pavel Soucek *

SUBJECTIVE TESTING AND OBJECTIVE MODELLING OF INFLUENCE OF DIFFERENT SOCIAL CLASSES TO VOICE CALL QUALITY PERCEPTION

The paper presents the analysis of different perception of quality of transmitted voice by different social classes in conference call. Two different social classes (namely people with great income, who are older than 35 and people with low income, who are younger than 25) have been examined by means of ITU-T P.800 conversational tests. Significant differences in perception between those two classes have been identified.

Keywords: *speech quality, subjective testing, Mean Opinion Score.*

1 Introduction

Modern telecommunication technology introduces several impairments into the voice transmission channel. Nowadays it is not enough to measure only physical parameters of the channel but it must be assessed how it affects resulting voice transmission quality.

To assess the quality of voice transmission the scale MOS (mean opinion score) is used (Table. 1). The term MOS is defined in Recommendation ITU-T P.800 [1].

MOS scale Tab. 1

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

There are several methods to obtain MOS values. The first and most accurate one is subjective testing, where the MOS value is obtained directly from users. Subjective tests are standardized by ITU in order to ensure repeatability of experimental results in [1].

Subjective tests are divided into the conversational and listening test. Conversational tests are more demanding on time and organization than listening. Therefore they are used mainly for testing the parameters of the transmission channel, which cannot be tested via a simple listening test, such as delay.

The second way to obtain MOS values is objective testing, which can be divided into two categories – intrusive and non-intrusive.

Intrusive methods usually deliver results nearest to subjective tests. They are based on a comparison of the original and transferred sample. These tests are based on algorithms, which use psychoacoustic models of human perception. They attempt to mathematically describe the human perception of sound and find variables which have a direct impact on the perceived quality of voice signal. Intrusive methods include several standardized algorithms, such as PESQ (Perceptual Evaluation of Speech Quality – ITU-T P.862 [2]) and POLQA (Perceptual Objective Listening Quality Assessment ITU-T P.863 [3]).

Another type of objective measurements is non-intrusive method. These methods do not use the reference signal and final MOS is calculated only from the parameters of the transferred sample. The disadvantage of these methods is a lower accuracy and reliability than in the case of intrusive methods. Example of non-intrusive method is 3SQM (Single Side Speech Quality Measurement – ITU-T P.563 [4]).

2 Work Performed

There are many impairment types that can affect conversation experience in telecommunications. One of the most significant is delay, because it directly influences user's experience. As we said

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earlier effects of delay can be assessed by means of conversational tests. We focused on teleconference calls, which are used for business calls, or online lessons. Usually they are used when participants are far apart. Because there are recommendations only for conversation between two users, we tried to extend methods from [1] for conversation with three participants.

The relationship between delay and resulting MOS score is known (i.e. [5]) and is reflected by many algorithms. The way how delay affects users depends on many factors. We believe that among others it is socio-economic background of user.

Tests were performed in Czech language.

2.1 Test-bed

Experiment was conducted in four separate rooms (Fig. 1) to avoid direct contact between the respondents. One of the rooms fully meets the requirements of the standard (reverberation 182 ms, −60 dB). The second room is particularly suited for the listening tests, meets the requirements of reverberation time <500ms, the other parameters have not been measured in the room. The third room is undefined in terms of acoustic parameters, we can assume that even this room meets the reverberation time <500ms. In the fourth room the technical background of the experiment is situated, the network simulator and posts of experiment supervisor.

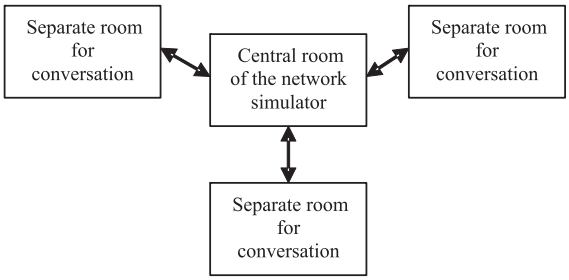

Fig. 1 Test-bed

2.2 Network Simulator

Respondent's posts include telephone chassis with standard handset. The signal from a handset microphone is pre-processed and routed into the central part of the simulator (Fig. 2). In the opposite direction is carried signal to the loudspeaker. The signal from the microphone is pre-processed in microphone amplifier (SHARK). The central part of the simulator consists of two digital signal processors. The first processor (DCX A) made a filtration with a Butterworth high-pass filter 48th level, 303Hz and low-pass filter Bessel 24th level, 3031Hz. Furthermore, the DCX A sets the first part of the variable delay in the range of 1-582 ms.

The processors are connected so that each of the three inputs of DCX B is the sum of the two different analogue outputs of DCX A. In the DCX the second part of the delay B is implemented and output signals are carried into the handsets of the respondents.

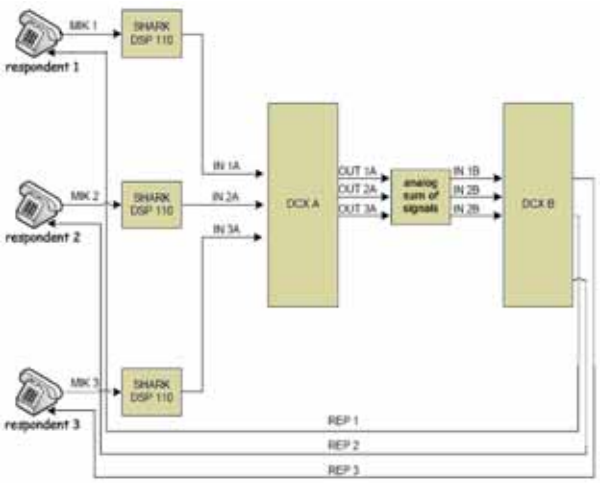

Fig. 2 Block diagram of network simulator

2.3 Delay

Delay is defined as time needed for voice signal to travel from talker to listener. Delay of telephone call in an IP network has several different causes. On the side of speaker it is particularly encoding, packetisation and controller interface. On the side of listener it is buffer, depacketisation and decoding. Causes of delays in the IP network itself are in particular: limited speed of signal transmission in the network and signal processing time of involved components such as routers and converters. The speed of the signal transmission is a particular problem when the call is made for long distance or part of the route led via satellite. In this experiment the following values of delay were adjusted: 62, 337, 612, 887 and 1176 ms.

2.4 Selection of Participants

Certain criteria must be met in selection of participants. They are described in detail in [1]. Among others participants should not be experts in area of telecommunication and they should have no hearing impairments.

As it was proven in previous experiments described in (i.e. [6]), participants are not able to distinguish between individual values of delay sometimes. Therefore all participants were instructed prior to testing that they should focus on delay.

Because of nature of our experiment we need participants with different socio-economic background. We decided to divide participants into two groups named Managers and Students.

Managers are people with higher education, with prestigious job position, above average income and their age is higher than 35. They are used to certain standards and they are willing to pay for quality. Also they expect to get quality they paid for.

In contrast, Students have lower income than managers and are used to get by with cheaper services. It was not necessary for participants from this group to be actually studying at the time, when tests took place.

In our experiment participated 55 people – 43 in group of Students, 9 in group of Managers. Other 3 participants were part of pre-test session, which was used for selection of conversation scenarios and proper way instruction session. The numbers of participants in both groups clearly show that Managers are much more difficult to acquire for participation in subjective tests.

2.5 Conversation Scenarios

There are several ways to conduct conversational tests. We used loosely defined scenarios based on real everyday life situations, which were from 2 to 3 minutes long. We tried to find scenarios which will be interactive enough, symmetric if possible. In the end we picked 5 following scenarios:

- Selecting gift – selecting present for friend, every participant have different budget and preferences
- Work on weekend – unexpected emergency work on weekend, participants already had plans for
- Party – participants are organizing party,
- Sport – participants have to decide which sport they will play, they have different preferences
- Culture event – participants have to decide which culture event they will attend, they have different preferences

It is clear from scenarios description that instructions for participants consist of two parts – common for all 3 participants and individual for each of them.

3 Results

In this part of paper we will evaluate whether there is difference between our two socio-economic groups. We will compare them directly and with results from objective tests.

3.1 Result Evaluation

Results from subjective test were processed and confidence intervals CI95 were computed using following formula:

$$u = \sqrt{\frac{1}{N(N-1)}\sum_{i=1}^{N}\left(MOS_i - \overline{MOS}\right)^2}$$

where N is number of scores for conversation and set delay
MOS_i is score from participants
$\overline{MOS}$ is average MOS

For CI95 for difference between Managers and students we used formula for indirect measurement:

$$u_{undirect} = \sqrt{\sum_{j=1}^{m}\left(\frac{\partial f}{\partial X_j}u_{x\_j}\right)^2}$$

where f is function used for computation of result (in our case simple difference)
X_j is one of the values (in our case *MOS* from one of our groups)
u_{x_j} is uncertainty of X_j

For comparison of subjective and objective tests we used several criteria.

Pearson's correlation coefficient:

$$r = \frac{\sum_{i=1}^{N}(X_i - \overline{X})(Y_i - \overline{Y})}{\sqrt{\sum_{i=1}^{N}(X_i - \overline{X})^2(Y_i - \overline{Y})^2}},$$

where $\overline{X}$, $\overline{Y}$ are mean values of measured samples
X_i, Y_i are the values of the i-th samples

Root mean squared error (RMSE):

$$RMSE = \sqrt{\frac{1}{N}\sum_{i=1}^{N}(MOS\_CQS_i - MOSC\_CQO_i)^2},$$

where N is number of tested delays
MOS_CQS_i is *MOS* result obtained by means of subjective testing
MOS_CQO_i is *MOS* result obtained by means of objective testing

Modified RMSE (RMSE*):

$$RMSE^* = \sqrt{\frac{1}{N}\sum_{i=1}^{N}(\Delta_i)^2},$$

where N is number of tested delays
Δ_i is 0 if difference between objective and subjective results is lesser than CI95 for subjective result and in other case it is given by following formula:

$$\Delta_i = \left|MOS\_CQS_i - MOSC\_CQO_i\right| - u_{k=2}$$

Maximum absolute difference:

$$D_{MAX} = MAX\left(\left|MOS\_CQS_i - MOS\_CQO_i\right|\right),$$

where MOS_CQS_i is *MOS* result obtained by means of subjective testing
MOS_CQO_i is *MOS* result obtained by means of objective testing

3.2 Subjective Tests

As we can see in Figs. 3 and 4 there is difference between our two groups. In case of both groups the quality drops with bigger delay. It is also clear that confidence intervals are bigger in case of Managers due to limited number of participants in this group.

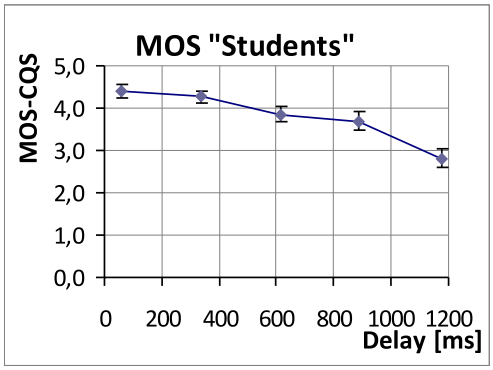

Fig. 3 MOS values for the group Students

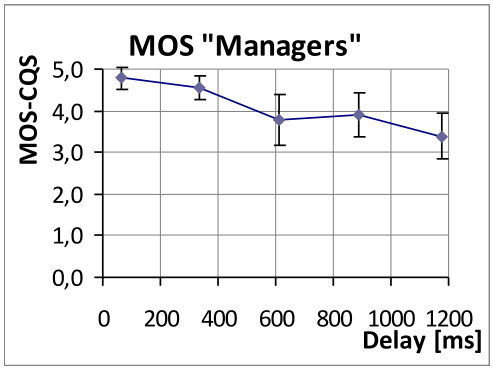

Fig. 4 MOS values for the group Mangers

Surprisingly from Fig. 5 is clear that the Managers are more tolerant to delay than the Students. This finding is directly opposite to our presumption that the Managers should be more demanding.

3.3 Objective Tests

From collected subjective date we created correction function. As we can see from Table 2, difference between group of Students and results obtained from PESQ are after proper regressions very similar. RMSE* was zero for both groups. Before correction it was 0.025 for Managers and 0 for Students.

4 Conclusion and Future Work

We have proven that there actually is difference between groups of users with different socio-economic background. It is surprising

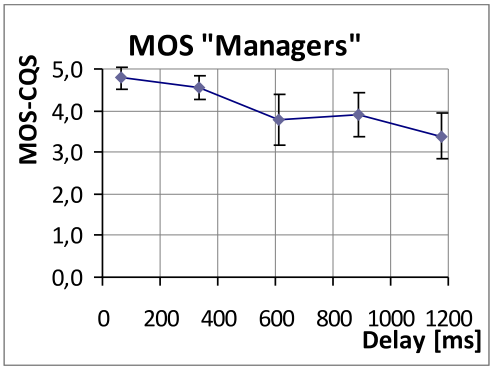

Fig. 5 Difference between the groups Managers and Students and its CI95. Important CI95 non-crossings with zero are obvious for the first two and last measured points.

Comparison between groups of managers and students Tab. 2

	Objective – "Managers"	Objective – "Students"
Criteria	Subjective – "Managers"	Subjective – "Students"
Pearson's correlation	0.952	0.990
RMSE	0.157	0.078
Maximum absolute difference	0.252	0.124

that the Managers, who we assumed should be used to higher standard, are less demanding. We suppose that this can be due to their higher experience with teleconference calls, or even due to the fact that they are usually older than participants from group of the Students, so they remember older technologies with less quality and are used to communication for longer distances. This was proven in [7]. On the other hand in [8] was proven that MOS scale shift during time even for same group of listeners.

For the future work we consider the validation of our results with higher number of participants from group of the Managers in order to reduce CI95.

5 Acknowledgements

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Oleg Vyacheslavovich Chernoyarov – Martin Breznan *

OPTIMAL AND QUASIOPTIMAL ALGORITHMS OF DISTINCTION OF THE COMPRESSED IMAGES IN BASES OF ORTHOGONAL POLYNOMIALS

The synthesis and analysis of effective practically realized algorithms of distinction of signals and images represented in bases of orthogonal polynomials was executed. Influence of distortions at formation of sensed and reference useful signals was considered. Recommendations about application of the synthesized distinction algorithms were formulated.

Keywords: *Signal and image distinction, generalized signal spectrum, maximum - likelihood receiver, incomplete signal reception probability of signal distinction.*

1. Introduction

Modern processing systems of the graphic and multimedia information (supervision of various objects, remote monitoring, control and aiming, etc.) should solve a problem of the operative analysis of complex dynamic images and transmission of information flows of a various type. In all these information structures the procedures of signal compression, signal restoration, parameters' measurement, recognition, etc are present. In a number of papers [1–3] it is shown that for the decision of many problems mentioned above the application of transformation of the initial image on bases of orthogonal polynomials or the functions connected with them is effective. In the present work the possibility of distinction of the images transformed thereby with use of the theory of statistical decisions is considered.

2. Problem statement

Let's assume that the realization of the two-dimensional random field $\xi(r)$, $r = (x, y)$, which can be or an additive mix of the useful signal $s_1(r)$ and hindrance $\eta(r)$: $\xi(r) = s_1(r) + \eta(r)$ (hypothesis H_1) or an additive mix of the useful signal $s_2(r)$ and hindrance $\eta(r)$: $\xi(r) = s_1(r) + \eta(r)$ (hypothesis H_2), incomes to the receiver input. Herewith the hypothesis H_1 is realized with probability p_1, and the hypothesis H_2 is realized with probability p_2.

Let's consider that initial useful signals $s_1(r)$, $s_2(r)$ can be expand into series of kind

$$s_1(r) = \sum_{m=0}^{\infty}\sum_{n=0}^{\infty} A_{mn}\varphi_{mn}(r), \qquad s_2(r) = \sum_{m=0}^{\infty}\sum_{n=0}^{\infty} B_{mn}\varphi_{mn}(r) \tag{1}$$

in one or another system of orthogonal functions $\{\varphi_{mn}(r)\}$. Here

$$A_{mn} = \int_{\Omega} s_1(r)\varphi_{mn}(r)dr, \qquad B_{mn} = \int_{\Omega} s_2(r)\varphi_{mn}(r)dr \tag{2}$$

are expansion factors, and Ω is area of orthogonality of basis $\{\varphi_{mn}(r)\}$.

It should be noted that useful signals in the process of transformation, compression and the subsequent restoration can be exposed to various restrictions. Firstly, the transmitting device can transmit signals with the limited number of modes M ($M < \infty$) to the communication channel:

$$s_1(r) \approx s_{1M}(r) = \sum_{m,n=0}^{M} A_{mn}\varphi_{mn}(r), \qquad s_2(r) \approx s_{2M}(r) = \sum_{m,n=0}^{M} B_{mn}\varphi_{mn}(r) \tag{3}$$

Secondly, depending on capacity of computing systems as a part of the receiver-analyzer, computational burden (computer timetable) etc not all modes of signals (1) but only the most powerful ones can be processed. In both cases so-called incomplete processing of a useful signal takes place.

Models of hindrances at optical and optoelectronic processing of the information are in detail considered in [4, 5]. Generally the additive hindrance hits in information system both at a stage of transformation of analogue signals $s_1(r)$, $s_2(r)$ in the discrete form and from the communication channel to the receiver-analyzer input. If a quantum character of weak optical signals is not taken into consideration and signals is assumed as intensive enough then it

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is possible often to believe, that a hindrance is Gaussian random field with known mathematical expectation $M\eta = \langle \eta(r) \rangle$ and correlation matrix $K\eta(r_1, r_2) = \langle \mathring{\eta}(r_1)\, \mathring{\eta}(r_2) \rangle$. Here $\mathring{\eta}(r) = \eta(r) - M\eta$ is a centered random field. As the mathematical expectation $M\eta$ is a priori known than it can be subtracted from a hindrance $\eta(r)$ in the processing operations and it can not be taken into consideration at the analysis. Thereupon a symbol «$\circ$» over $\eta(r)$ for simplification of record will be skipped further.

Spectral-correlation properties of a field $\eta(r)$ can be various depending on physical statement of a problem [4, 5]. If processing algorithms don't consider thin difference of spatial spectrums of a useful signal and a hindrance but focus the attention on the analysis of generalized spectrum's modes of the field $s_i(r)$, $i = 1, 2$ than hindrance $\eta(r)$ may be approximated by Gaussian white noise with a correlation matrix of a kind $K\eta(r_1, r_2) = (N_0/2)\delta(r_2 - r_1)$. The model of white noise is quite proved if hindrance sources are wideband processes in electronic devices [6].

3. Optimal and Quasioptimal Algorithms of Images' Distinction in Bases of Orthogonal Polynomials

According to [7] the algorithm of distinction of signals $s_i(r)$, $= 1, 2$ on the background of noise $\eta(r)$ can be presented in the general form as follows:

$$L \underset{<}{\overset{>}{}} h_0\,, \tag{4}$$

where L is a logarithm of functional of likelihood ratio (FLR) for the hypothesis H_1 against alternative H_2, and h_0 is the threshold calculated according to chosen optimality criterion. If logarithm of FLR exceeds the threshold h_0 then the decision on signal presence $s_1(r)$ in realization of the observable data $\xi(r)$ is delivered and otherwise – on signal presence $s_2(r)$.

Believing a hindrance $\eta(r)$ by Gaussian white noise for logarithm of FLR we have [7]

$$L = \frac{2}{N_0}\int_\Omega \xi(r)\big[s_1(r) - s_2(r)\big]dr - \frac{E_1 - E_2}{N_0}\,. \tag{5}$$

Here $E_i = \int_\Omega s_i^2(r)dr$, $i = 1, 2$ is a total energy of signal $s_i(r)$. Then the decision rule (4) will be presented in the form

$$\frac{2}{N_0}\int_\Omega \xi(r)\big[s_1(r) - s_2(r)\big]dr - \frac{E_1 - E_2}{N_0} \underset{<}{\overset{>}{}} h_0\,. \tag{6}$$

At incomplete reception when the mix $\xi(r) = s_{iM}(r) + \eta(r)$ with unknown number of modes M at a useful signal (3) is observed for the hypothesis H_i we receive from Eq. (6)

$$L = \frac{2}{N_0}\int_\Omega \xi(r)\big[s_1(r) - s_2(r)\big]dr - \frac{E_1 - E_2}{N_0} \underset{<}{\overset{>}{}} h_0\,. \tag{7}$$

According to [7] the detection algorithm (7) will not be optimal now. We name Eq. (7) as quasioptimal decision rule. Really, Eq. (7) turned into optimal decision rule (6) at $M \to \infty$. Comparing characteristics of algorithms (6) and (7) it is possible to estimate losses in efficiency of processing algorithm (7) and to define requirements to throughput of a communication channel.

If ν modes for representation of reference signals $s_1(r)$, $s_2(r)$ on the receiving site is used then the decision rule (6) will be transformed to form

$$L = \frac{2}{N_0}\int_\Omega \xi(r)\big[s_{1\nu}(r) - s_{2\nu}(r)\big]dr - \frac{E_{1\nu} - E_{2\nu}}{N_0} \underset{<}{\overset{>}{}} h_0 \tag{8}$$

where $E_{i\nu} = \int_\Omega s_{i\nu}^2(r)dr$, $i = 1, 2$. Signals $s_{i\nu}(r)$ in Eq. (8) containing ν modes are defined similarly Eq. (3).

Generally the number ν can be both more and less M. It is determined by restriction on computing possibilities of the receiver-analyzer, the prior information on channel throughput and other restrictions on structure of information transmission and processing system.

In a number of practical applications there can be a notation of Eqs. (4), (5) in a vector-matrix form more convenient. We will designate

$$X_{mn} = \int_\Omega \xi(r)\varphi_{mn}(r)dr\,,$$

$$C_{mn}^- = A_{mn} - B_{mn},\ C_{mn}^+ = A_{mn} + B_{mn}.$$

Then it is possible to write down Eqs. (4), (5) as

$$L = \frac{1}{N_0}\sum_{m,n=0}^{\infty}(2X_{mn} - C_{mn}^+)C_{mn}^- \underset{<}{\overset{>}{}} h_0\,.$$

The similar notation can be offered for algorithms (7), (8).

4. Efficiency of Image Distinction Algorithms in Bases of Orthogonal Polynomials

As the quantitative characteristic of image distinction algorithms synthesized in item 3 the mean error probability will be used [7]. We will find the error probability of signal distinction at optimal reception according to algorithm (6) at first. Following [7] the mean error probability of distinction P_e we will write down in a form

$$P_e = p_1 P(2|1) + p_2 P(1|2), \tag{9}$$

where p_i is the prior probability of the hypothesis H_i and $P(i|j)$ is the probability of decision in favour of i-th signal while j-th hypothesis was true ($i, j = 1, 2$). By definition [7]

$$P(2|1) = \int_{-\infty}^{h_0} w_1(x)dx\,,\ P(1|2) = \int_{h_0}^{\infty} w_2(x)dx\,. \tag{10}$$

Here $w_i(x)$, $i = 1, 2$ is the probability density of logarithm of FLR (5) at hypothesis H_i.

According to [7] logarithm of FLR L (5) is a Gaussian random value. Then for probabilities (10) is:

$$P(1|2) = 1 - \Phi\left[(h_0 - m_{2L})/\sigma_{2L}\right],$$
$$P(2|1) = \Phi\left[(h_0 - m_{1L})/\sigma_{1L}\right], \quad (11)$$

Through m_{1L}, m_{2L} and σ_{1L}, σ_{2L} in Eq. (11) mathematical expectations and mean square deviations of random value L (5) at hypotheses H_1 and H_2 are designated. Implementing immediate averaging of Eq. (5) on all possible realizations of the observable data $\xi(r)$ we can find following expressions for m_{1L}, m_{2L}, σ_{1L}, σ_{2L}:

$$m_{1L} = -m_{2L} = (E_1 + E_2 - 2R\sqrt{E_1E_2})/N_0,$$
$$\sigma^2_{1L} = -\sigma^2_{2L} = 2(E_1 + E_2 - 2R\sqrt{E_1E_2})/N_0. \quad (12)$$

Here is the correlation factor reflecting similarity of the geometrical form of two objects.

Let's introduce designations $m = (E_1 + E_2 - 2R\sqrt{E_1E_2})/N_0$, $D = 2(E_1 + E_2 - 2R\sqrt{E_1E_2})/N_0$. Then, substituting (12) in (11), and (11) in (9), for the mean error probability of distinction we can write down

$$P_e = p_1\left[1 - \Phi\left(\frac{h_0 + m}{\sqrt{D}}\right)\right] + p_2\Phi\left(\frac{h_0 + m}{\sqrt{D}}\right) \quad (13)$$

The formula (13) becomes simpler if signals $s_1(r)$ and $s_2(r)$ are equiprobable (prior probabilities of hypotheses H_1 and H_2 are equal): $p_1 = p_2 = 0{,}5$ and the threshold according to criterion of the ideal observer [7] is chosen ($h_0 = 0$):

$$P_e = 1 - \Phi\left(\frac{m}{\sqrt{D}}\right) = 1 - \Phi\left(\frac{\sqrt{z_1^2 + z_2^2 - 2z_1z_2R}}{2}\right) \quad (14)$$

Here $z^2_{1,2} = 2E_{1,2}/N_0$. If in addition to the discriminating signals are symmetric, i.e. $E_1 = E_2 = E$ ($z_1 = z_2 = z$) than

$$P_e = 1 - \Phi\left[z\sqrt{(1 - R)/2}\right]. \quad (15)$$

Let's come to a case of distinction of two signals (3) limited on number of modes now. But formed reference signal assumes processing of infinite number of modes of the observable field $\xi(r)$. If representation (3) is performed then logarithm of FLR (7) will be Gaussian random value with parameters

$$m_{1L} = \frac{2}{N_0}\left(\sum_{m,n=0}^{M} A^2_{mn} - \sum_{m,n=0}^{M} A_{mn}B_{mn}\right) - (E_1 - E_2)/N_0 \quad (16)$$

$\sigma^2_{1,L}$ (12) at the hypothesis H_1 and

$$m_{2L} = \frac{2}{N_0}\left(\sum_{m,n=0}^{M} A_{mn}B_{mn} - \sum_{m,n=0}^{M} B^2_{mn}\right) - (E_1 - E_2)/N_0 \quad (17)$$

$\sigma^2_{2,L}$ (12) at the hypothesis H_2.

We designate

$$\varepsilon_{1M} = \sum_{m,n=0}^{M} A^2_{mn} \Big/ \sum_{m,n=0}^{\infty} A^2_{mn},$$

$$\varepsilon_{2M} = \sum_{m,n=0}^{M} B^2_{mn} \Big/ \sum_{m,n=0}^{\infty} B^2_{mn},$$

$$R_M = \varepsilon_{1M} = \sum_{m,n=0}^{M} A_{mn}B_{mn} \Big/ \sqrt{\sum_{m,n=0}^{\infty} A^2_{mn} \sum_{m,n=0}^{\infty} B^2_{mn}},$$

take into account that $E_1 = \sum_{m,n=0}^{\infty} A^2_{mn}$, $E_2 = \sum_{m,n=0}^{\infty} B^2_{mn}$ and overwrite Eqs. (16), (17) as

$$m_{1L} = \left[E_1 + E_2 - 2R_M\sqrt{E_1E_2} - 2E_1(1 - \varepsilon_{1M})\right]/N_0 \quad (18)$$

$$m_{2L} = -\left[E_1 + E_2 - 2R_M\sqrt{E_1E_2} - 2E_2(1 - \varepsilon_{2M})\right]/N_0$$

Then according to Eq. (9) the mean error probability P_e of distinction of two signals (3) will be defined as follows

$$P_e = p_1\left[1 - \Phi\left(\frac{h_0 - m_{2L}}{\sigma_{2L}}\right)\right] + p_2\Phi\left(\frac{h_0 - m_{1L}}{\sigma_{1L}}\right). \quad (19)$$

The formula (19) becomes simplest if $h_0 = 0$, $p_1 = p_2 = 0{,}5$, $E_1 = E_2 = E$, $\varepsilon_{1M} = \varepsilon_{2M} = \varepsilon_M$. In this case

$$P_e = 1 - \Phi\left[z(\varepsilon_M - R_M/\sqrt{2(1 - R)}\right], \quad (20)$$

where $z^2 = 2E/N_0$. At $M \to \infty$ we have: $\varepsilon_M \to 1$, $R_M \to R$ and expression (20) proceed to (15). It may be noted that performance degradation of distinction (20) in comparison with a case of optimal reception (15) takes place if

$$\sqrt{1 - R} > (\varepsilon_M - R_M)/\sqrt{1 - R} \text{ or}$$
$$1 - R - (\varepsilon_M - R_M) > 0. \quad (21)$$

From a Cauchy-Bunyakowsky-Schwartz inequality [8] follows that

$$\sum_{m,n=0}^{M} A_{mn}B_{mn} \le \sqrt{\left(\sum_{m,n=0}^{M} A^2_{mn}\right)\left(\sum_{m,n=0}^{M} B^2_{mn}\right)} \text{ or } R_M \le \varepsilon_M.$$

Consequently the inequality (21) may be broken for strongly correlated images, when $R \to 1$, and distinction with use of limited number of modes becomes more preferable than with use of unlimited number of modes.

Consider an illustration of the offered approach to distinction of signals/images by giving an example. For simplification of mathematical calculations and visualization of received results we will believe that discriminated signals are one-dimensional. For two-dimensional signals the general conclusions will be similar.

For definiteness we will assume that

$$s_1(x) = \exp(-\alpha^2x^2), \quad (22)$$

$s_2(x) = s_1(x - \tau)$ and correlation factor R (12) is $R = R(\tau) = \exp(-\alpha^2\tau^2/2)$. Here α is the parameter characterizing the duration of discriminated signals. As system of orthogonal functions [10] we will choose Hermitian functions [9]:

$$\varphi_n(t) = \frac{1}{\sqrt{2^n n!\sqrt{\pi}}} H_n(t)\exp\left(-\frac{t^2}{2}\right).$$

Here $H_n(t)$ is a Hermitian polynomial of n-th order [9]. Then for the generalized spectrums A_n, B_n (2) of signals $s_1(x)$, $s_2(x)$ it can be found:

$$A_n = \frac{(-1)^{n/2}\sqrt[4]{\pi}}{(n/2)!}\sqrt{\frac{n!}{2^{n-1}(2\alpha^2+1)}}\left(\frac{2\alpha^2-1}{2\alpha^2+1}\right)^{n/2},$$

$$B_n = B_n(\tau) = \frac{\sqrt[4]{\pi}}{\sqrt{2^{n-1}n!(2\alpha^2+1)}}\left(\frac{2\alpha^2-1}{2\alpha^2+1}\right)^{n/2}\times$$

$$\times\exp\left(-\frac{\alpha^2\tau^2}{2\alpha^2+1}\right)H_n\left(\frac{2\alpha^2\tau}{\sqrt{4\alpha^4-1}}\right).$$

In Fig. 1 we show the dependences of the mean error probability of distinction P_e of signals $s_1(x)$ and $s_2(x)$ calculated with using Eq. (18), (19) for $h_0 = 0$, $p_1 = p_2 = 0{,}5$, $\alpha = 1$, $\tau = 2$. Curve 1 corresponds to a processing of first two modes of signals ($M = 1$), curve 2 to a processing of first three modes ($M = 2$) and curve 3 to a processing of first five modes ($M = 4$). Here the limiting values of probability P_e (15) at optimal reception ($M = \infty$) by circles are also marked. From Fig. 1 it is obvious that the account of the several first modes of expansion of useful signals $s_1(x)$ and $s_2(x)$ provides the characteristics of distinction close to limiting.

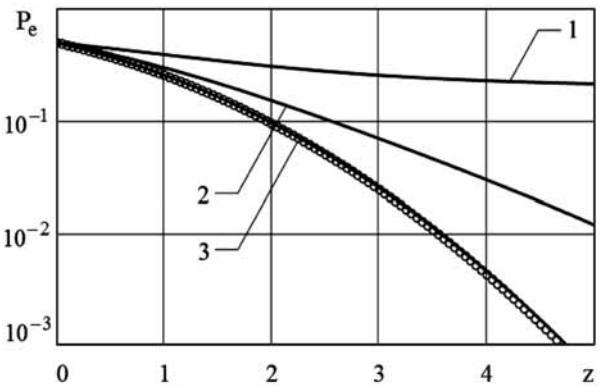

Fig. 1 Dependences of the mean error probability of distinction P_e of signals $s_1(x)$ and $s_2(x)$

Let's put now that the reference signal of the optimal receiver contains also the limited number of modes, i.e. the structure of distinction algorithm is defined by a rule (8) and M > ν. In this case the logarithm of FLR (8) is a Gaussian random value with mathematical expectation and variance

$$m_{1L} = (E_{1\nu} + E_{2\nu} - 2R_\nu\sqrt{E_{1\nu}E_{2\nu}})/N_0,$$
$$\sigma_{1L}^2 = 2(E_{1\nu} + E_{2\nu} - 2R_\nu\sqrt{E_{1\nu}E_{2\nu}})/N_0 \quad (23)$$

at the hypothesis H_1 or

$$m_{2L} = -(E_{1\nu} + E_{2\nu} - 2R_\nu\sqrt{E_{1\nu}E_{2\nu}})/N_0,$$
$$\sigma_{2L}^2 = 2(E_{1\nu} + E_{2\nu} - 2R_\nu\sqrt{E_{1\nu}E_{2\nu}})/N_0 \quad (24)$$

at the hypothesis H_2. Here $R_\nu = \int_\Omega s_{1\nu}(r)s_{2\nu}(r)dr/\sqrt{E_{1\nu}E_{2\nu}}$.

From Eqs. (23), (24) it is easy to see that

$$m_{1L} = -m_{2L} = m_\nu,\ \sigma_{1L}^2 = \sigma_{2L}^2 = \sigma_\nu^2,\ \sigma_\nu^2 = 2m_\nu.$$

Then for the mean error probability of distinction P_e we can write down similarly Eq. (19):

$$P_e = p_1\left[1 - \Phi\left(\frac{h_0}{\sqrt{2m_\nu}}\right) + \sqrt{\frac{m_\nu}{2}}\right] +$$
$$+ p_2\Phi\left(\frac{h_0}{\sqrt{2m_\nu}} - \sqrt{\frac{m_\nu}{2}}\right) \quad (25)$$

The formula (25) becomes simplest at $p_1 = p_2 = 0{,}5$ and $h_0 = 0$:

$$P_e = 1 - \Phi\left(\sqrt{\frac{E_{1\nu} + E_{2\nu} - 2R_\nu\sqrt{E_{1\nu}E_{2\nu}}}{2N_0}}\right) =$$
$$= 1 - \Phi\left(\frac{\sqrt{z_1^2\varepsilon_{1\nu} + z_2^2\varepsilon_{2\nu} - 2z_1z_2R_\nu\sqrt{\varepsilon_{1\nu}\varepsilon_{2\nu}}}}{2}\right). \quad (26)$$

If, besides, $\varepsilon_{1\nu} = \varepsilon_{2\nu} = \varepsilon_\nu$, $z_1 = z_2 = z$ then

$$P_e = 1 - \Phi\left[z\sqrt{\varepsilon_\nu(1 - R_\nu)}/2\right]. \quad (27)$$

In Eqs. (26), (27) it is designated $\varepsilon_{i\nu} = E_{i\nu}/E_i$, $i = 1, 2$, and variable z_i is defined the same as in Eq. (14). From comparison of Eqs. (15) and (27) it is possible to conclude that incomplete reception (because of restrictions in mode structure of the receiver's reference signal. i.e., as consequence, condition performance $\varepsilon_\nu < 1$) bring loses of optimal reception generally.

In Fig. 2 the dependences $P_e = P_e(z)$ calculated with use of formula (26) for signal models $s_1(x)$ (22) and $s_2(x) = s_1(x - \tau)$

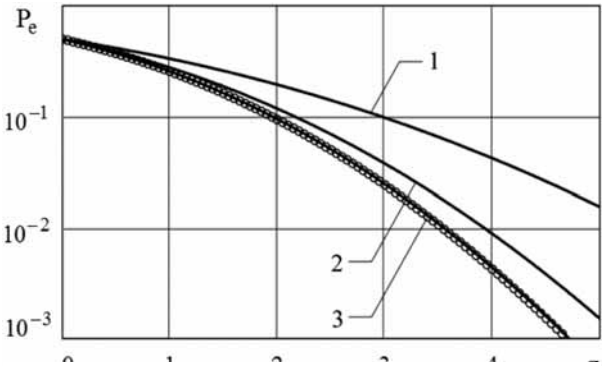

Fig. 2 Dependences $P_e = P_e(z)$ calculated with use of formula (26)

at $\alpha = 1$, $\tau = 1$ and various numbers of processed modes ν are shown. Curve 1 is plotted at $\nu = 1$ (the reference signal contains two modes), curve 2 at $\nu = 2$ (the reference signal contains three modes) and curve 3 at $\nu = 4$ (i.e., five modes of the reference signal are taken into account).

In Fig. 3 the similar dependences of probability P_e (19) and (26) are represented for following variants of incomplete reception.

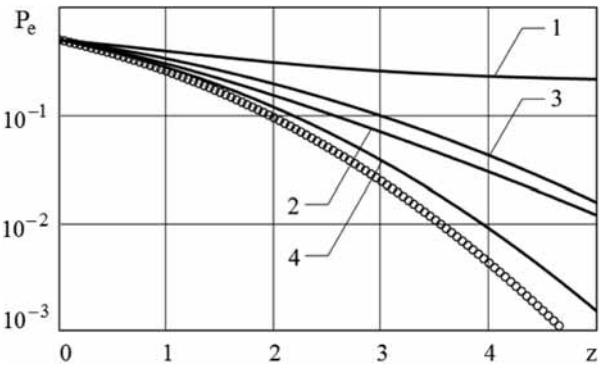

Fig. 3 Similar dependences of probability P_e (19) and (26)

Curves 1 and 2 show the mean error probability of distinction when the sensed signal has two or three modes. Curves 3 and 4 are plotted for a case of restriction of the reference signal consisting of two (curve 3) or three (curve 4) modes. Circles in Figs. 2, 3 map limiting values of the probability P_e (15) at optimal reception.

From graphs shown in Fig. 3 it is obvious that restrictions on number of modes in the reference signal are more advantageous energetically than restrictions in the sensed signal. This is due to the fact that if restrictions of the sensed signal occur then a loss of a part of useful signal energy takes place while noise energy remains invariable. And vise versa if restrictions of the reference signal occur that not only useful signal energy but also noise energy decreases on the receiver output.

5. Conclusion

At synthesis of distinction algorithms of signals and images their representation in the form of a set of the generalized spectrum's factors can appear rather effective. The given representation allows to receive simpler and practically realized discriminators of signals and images observed on the noise background. For strongly correlated signals and images the use of final mode number of sensed signals/images can provide smaller sacrifices of distinction quality in comparison with a case of distinction of full signals/images. An incomplete reception of a useful signal causes loses of optimal reception in general because of restrictions in mode structure of a reference signal. However, the mode restrictions in a reference signal are more advantageous energetically (at the expense of reduction not only energy of a useful signal but also noise energy on a receiver output) than corresponding restrictions in the sensed signal. In addition, by expanding of useful signal to cover several first modes, this process can provide the distinction characteristics close to limiting.

The offered processing algorithms of signals and images can be almost realized with use of modern element base on the basis of digital signal processors or programmed logic integrated chips.

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Muhammad Saeed Khan – Roberto Nebuloni – Martin Grabner – Carlo Capsoni – Erich Leitgeb – Vaclav Kvicera *

STATISTICAL CHARACTERIZATION OF AEROSOLS' LIQUID WATER CONTENT AND VISIBILITY FOR TERRESTRIAL FSO LINKS

Liquid water content (LWC) is a microphysical parameter which is used to characterize fog/cloud and other particulates. In the absence of precipitation higher amount of LWC depicts dense fog, significantly reducing visibility and causing performance degradation of free-space optical (FSO) links for a non-negligible amount of time. A measurement campaign in Prague (Czech Republic) has recorded the specific attenuation caused by fog in terrestrial FSO links along with visibility, liquid water content (LWC) and integrated particle surface area (PSA) for operational wave-lengths of 1550 nm and 830 nm (installed at two different path links). This contribution presents and analyzes 5 months real time measured data of LWC and visibility for terrestrial FSO links in an attempt to find the best fit distribution model for aerosol's liquid water content and visibility for terrestrial FSO links under reduced visibility conditions. The Probability Density Function (PDF) estimation of visibility and LWC puts forth the Gamma distribution as the best fit distribution for both. The results provide the statistical behaviour of LWC and visibility, which is of high importance for Optical Wireless systems and networks.

Keywords: *Free-space Optics (FSO), Optical Wireless Communication, Liquid Water Content (LWC), Visibility.*

1. Introduction

Free-space optical (FSO) links are supposed to operate through atmosphere which contains fog, smoke, dust, rain, smog and charged particles. The terrestrial FSO link provide a viable last mile solution for high speed connectivity without the need of digging necessary to lay the conventional fiber [1]. Atmospheric particles like fog, snow, etc attenuate the transmitted signal propagating through the line-of-sight FSO links. Among all the different attenuating factors, fog is the most serious deterrent [2, 3]. Fog can be characterized by liquid water content (LWC), optical visibility, drop size distribution and temperature [4]. The scattering, absorption and extinction of laser beam propagating through the atmosphere are associated with fog droplets sizes, their effective radii [3] and the microphysical properties of fog, specifically liquid water content [3, 5].

The LWC is the measure of total mass of water per unit volume of the droplets in air. The LWC is expressed in g/m^3. LWC is used to characterize the different types of fog/clouds. In the absence of precipitation higher amount of liquid water content decreases visibility and indicates dense fog [5, 6].

Fog is an accumulation of water droplets or ice crystal fines accompanied by hygroscopic, water-saturated fine particles that reduce visibility. Under dense fog conditions the visibility decreases below 100 m and cause sever signal attenuation for FSO links [2]. For light fog the visibility remains in the range of 500–1000 m relatively less deterrent for FSO links.

For wide spread acceptability of FSO links there is need to investigate thoroughly different weather impairment on FSO links [7]. The prediction of optical attenuation in lower atmospheric visibility ranges due to water hazes, fogs, and clouds has been thoroughly investigated and researched [8].

To the best of the authors' knowledge this is 1st attempt to find a distribution model for LWC for terrestrial FSO links. Fog drop size distribution is widely modelled as a modified gamma distribution [9, 10]. The current work found Gamma distribution as a best fit model for visibility from 5 months measured data. Curve fitting techniques have been used to find the best fit PDF for LWC and visibility. All the continuous distributions have been compared using the Quantile-Quantile Plot (Q-Q plot). Here we presented the two best fit distributions for the analysis. The current work is

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of high importance for optical wireless communication and networks and will provide a thorough understanding of visibility and LWC and their variations.

2. Experimental Setup

The measurement campaign was carried out at the Department of Frequency Engineering, Czech Metrology Institute (CMI) Prague, Czech Republic from 08. 01. 2009 till 31.05.2009. Prague is located at latitude 50° 05' 12" N longitude 14° 24' 59" E and at altitude of 191 m. Prague has continental weather. Prague has the air mean temperature 10.4° C from 1971–2000 [11]. The FSO systems were installed 26 meters above the ground level. The link margin of the two FSO systems allows the measurement of specific attenuation up to 180 dB/km for 830 nm wavelength systems and 130 dB/km for 1550 nm system. Optical calibration was performed before deploying the FSO devices. A received power is obtained from the calibrated Received Signal Strength Indicator (RSSI) signal of the FSO link. Meteorological conditions are identified by means of a color video camera and an automatic weather observation system located near the FSO receivers. The system uses Vaisala sensors for the measurement of temperature, humidity, air pressure, velocity and direction of the wind. The VAISALA PWD 11 equipment measures the atmospheric visibility (5% definition) values in the range from 50 m up to 2000 m using forward scattered light in the angle of 45°. The PVM-100 device is used to measure liquid water content LWC (g/m^3) and integrated particle surface area PSA (cm^2/m^3) of fog.

The meteorological data is synchronized in time with the hydrometer attenuation measurement. The received FSO signal levels and the meteorological data are recorded synchronously on a PC's hard disk. In this article we analyzed 5 months real time measurement data of reduced visibility and LWC due to fog for terrestrial FSO links. In the measurement campaign, we sampled data at rate of one sample per minute. The other instruments were calibrated in a way to measure the specific quantity at the last second of the every minute in parallel with the optical attenuations. We selected data set for analysis where the visibility was less than 1 Km. Figs 1 and 5 shows the reduced data set of LWC and Visibility. It is important to note that the time axis of Figs 1 and 5 is not continuous.

3. Results and Analysis

The department of Frequency Engineering at Czech Metrology Institute is conducting measurement campaign at path links of 100 m and 853 m using the wavelengths of 1550 nm, 830 nm and 850 nm along with visibility, LWC and integrated particle surface area (PSA). A 5 months measured data of visibility and LWC were collected for detailed analysis. In the measurement campaign we sampled data at a rate of one sample per minute. The other instruments were calibrated in a way to measure the specific quantity at the last second of the every minute in parallel with the optical attenuation. In metrology theory, it is admitted that fog exists when visibility is decreased to less to less than 1 km. Therefore we selected the data set for analysis where the visibility was less than 1 km. Fig. 1 shows the reduced data set. It is also important to remark that the time axis in the figure is not continuous. The detailed statistical analysis to find the best suited distribution model for visibility and LWC data is provided in subsequent sections.

A. Visibility

Visibility is defined as the distance to an object at which the image contrast drops to a certain percentage of the original contrast of the object, equivalent to a certain transmission threshold tTH over the atmospheric path. Two different definitions for threshold exist, 2% and 5%. The 5% transmission threshold is more commonly used at airports to denote the "runway visibility range" RVR [12, 13]. Another meteorological definition of visibility is the actual distance at which a person can discern an ideal dark object against the horizon sky [14]. The visibility range is technically measured at the center of sensitivity for the human eye (where the sensitivity of the human eye is maximum), at 550 nm with a spectral band width of typically 250 nm. Different visibility based empirical models have been proposed to calculate optical attenuation from visibility, Kim [15], Pierce [16], Kruse [17] and Al-Naboulsi [18]. Here we presented the statistical characterization of 5 months measured visibility data. The analysis of measured visibility is provided in Fig. 1.

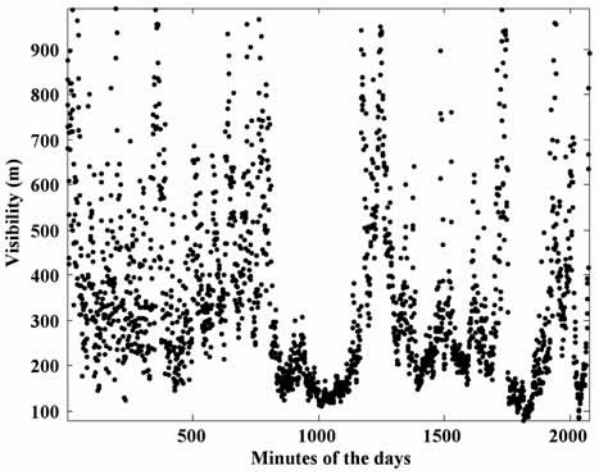

Fig. 1 Analysis of measured visibility

The horizontal axis of Fig. 1 shows the minutes of the days and the vertical axis shows the visibility measured in meters. It is clear from Fig. 1 that there occurred certain dense fog events when the visibility is less than 100 m. The descriptive statistics of the visibility is provided in Table l.

It is important to mention that the minimum value of measured visibility is 80 m (from Table 1) while the maximum value is 991 m. And also the skewness of the data is positive which shows that the right tail of the distribution will be longer as compared to the left tail.

Descriptive statistics of the measured visibility Tab. 1

Serial No	Statistical Quantity	Values
1	Sample Size	2073
2	Range	911
3	Mean	343.74
4	Variance	37688.0
5	Std. Deviation	194.13
6	Std. Error	4.2639
7	Skewness	1.1366
8	Min	80
9	25% (Q1)	197.5
10	50% (Median)	291
11	75% (Q3)	443
12	90%	637.2
13	95%	752.2
14	Max	991

After having explored the descriptive statistics we proceed further to find the best fit distribution for visibility measured data. We compared all the continuous distribution on the visibility data by analysing the probability density function (PDF) and Cumulative distribution function (CDF). We performed comparison among all the distribution by comparing Quantile-Quantile-Plot (QQ plot). QQ plot is used for comparing two probability distributions by plotting their quantiles against each other. Here we show the results of two best fitted distribution for visibility data. The PDF and CDF of the measured visibility data is provided in Figs 2 and 3 respectively.

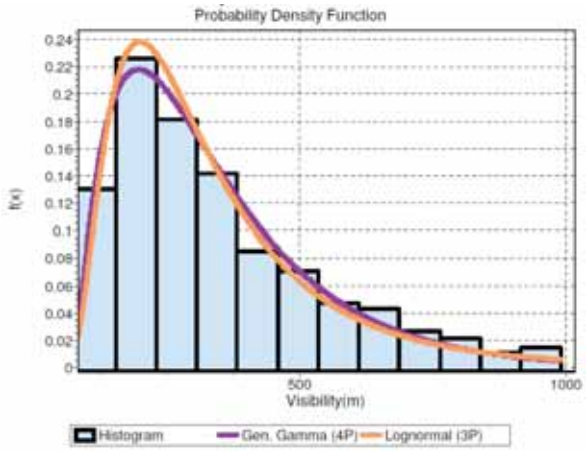

Fig. 2 PDF over the histogram of measured visibility data

Figs. 2 and 3 show that the selected distribution models can be used for statistical characterization of the behavior of measured visibility data. We performed the goodness of fit by observing their

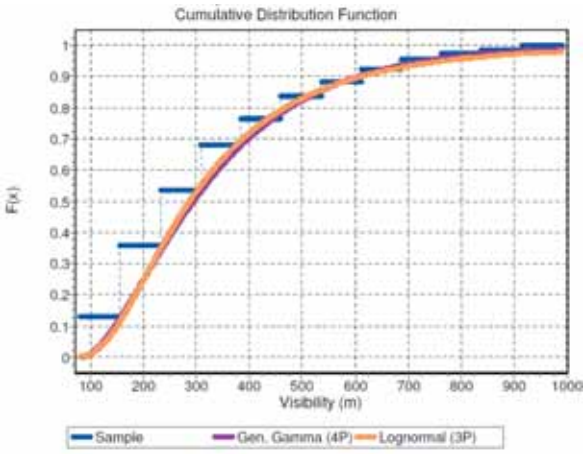

Fig. 3 CDF of the measured visibility

QQ plot of measured visibility to find the best fit model between the selected two models. The QQ plot of measured visibility is provided in Fig. 4. In Fig. 4 it is clear that Gamma distribution is performing well as compared to Lognormal distribution for measured visibility. The QQ plot for Gamma distribution is following the normal line as compared to that of Lognormal distribution.

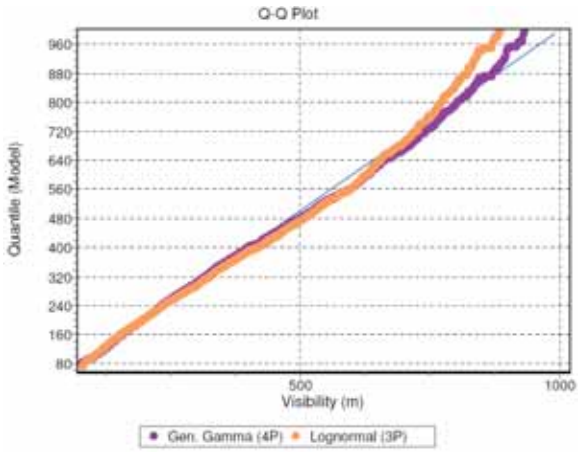

Fig. 4 QQ Plot for Visibility data

The probability density function for Gamma distribution is provided in equation (1).

$$f(x) = \frac{\kappa(x-y)^{\kappa\alpha-1}}{\beta^{\kappa\alpha}\Gamma(a)}\exp\left(-\left((x-y)/\beta\right)\kappa\right) \qquad (1)$$

Where κ is a continuous shape parameter ($\kappa > 0$), continuous shape parameter ($\alpha > 0$), β is scale parameter ($\beta > 0$) and γ is location parameter. We computed the optimum parameters for Gamma distribution for measured visibility which are provided in Table 2.

Optimum parameters for best fitted distribution Tab. 2

Description	Distribution Model	Parameters
Visibility	Gamma	$\kappa = 0.79766$, $\alpha = 2.8018$, $\beta = 69.203$, $\gamma = 78.538$
LWC	Gen. Gamma	$\gamma = 0.96962$, $\alpha = 1.4269$, $\beta = 0.0438$

We concluded that Gamma distribution (4 parameters Gamma Distribution model) is the best suited model for measured visibility in the absence of precipitation. The proposed distribution model can be used under fog conditions.

B. Liquid water content

LWC is the measurement of mass of water in fog/cloud in a specified amount of dry air. LWC can be expressed as g/m^3. LWC is different for different types of fog and clouds. The classification of clouds and fog is highly related to the amount of LWC and its origin. The combination of LWC and its origin allows to readily predict the types of conditions that will be, most likely, in the vicinity of the FSO links [19]. Fog that has very low densities contains very small amount of water and so eventually results in lower values of LWC i.e., about 0.05 g/m^3 for a moderate fog (visibility range around 300 m). Much higher values of LWC (around 0.5 g/m^3) result in the formation of thick or dense fog (visibility range of about 50 m) [10]. Fog characterized by several physical parameters like particle size distribution, temperature, humidity and LWC has been extensively modeled by drop size distribution and visibility range.

The analysis of the measured LWC is provided in Fig. 5.

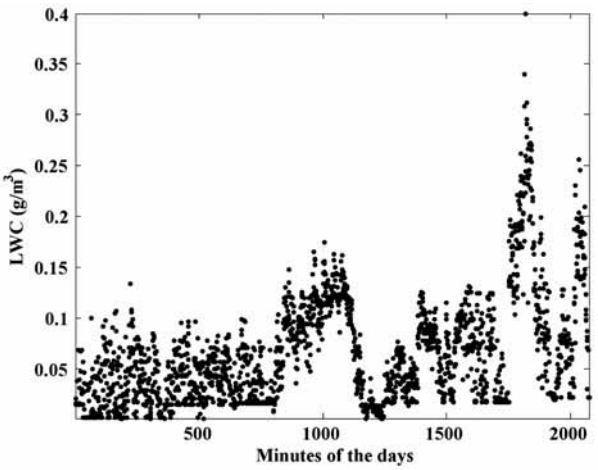

Fig. 5 Analysis of measured LWC

The horizontal axis of Fig. 5 shows the minutes of the days and the vertical axis shows the LWC measured in g/m^3. The descriptive statistics of the measured LWC is provided in Table 3.

Descriptive statistics of the measured LWC Tab. 3

Serial No	Statistical Quantity	Values
1	Sample Size	2073
2	Range	0.399
3	Mean	0.06422
4	Variance	0.00281
5	Std. Deviation	0.05304
6	Std. Error	0.00116
7	Skewness	1.4467
8	Min	0.001
9	25% (Q1)	0.021
10	50% (Median)	0.053
11	75% (Q3)	0.093
12	90%	0.128
13	95%	0.17
14	Max	0.4

It is obvious from Table 3 that 95 % of the time the LWC remains below 0.17 g/m^3.

We applied curve fitting techniques to find the best fit distribution model for measured LWC. We compared all the continuous distribution on the LWC data by analysing their PDF and CDF. We performed comparison among all the distribution by comparing the QQ plot. The PDF, CDF of the measured LWC data is provided in Figs. 6 and 7 respectively. Fig. 6 shows that both selected distribution models can be used for statistical characterization of LWC under fog conditions. But the results of QQ plot suggest that Gamma distribution is a better suited model for LWC than compared Beta distribution.

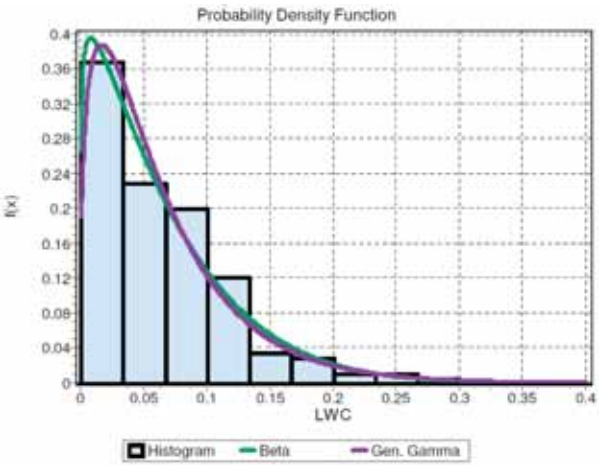

Fig. 6 PDF over the histogram of measured LWC

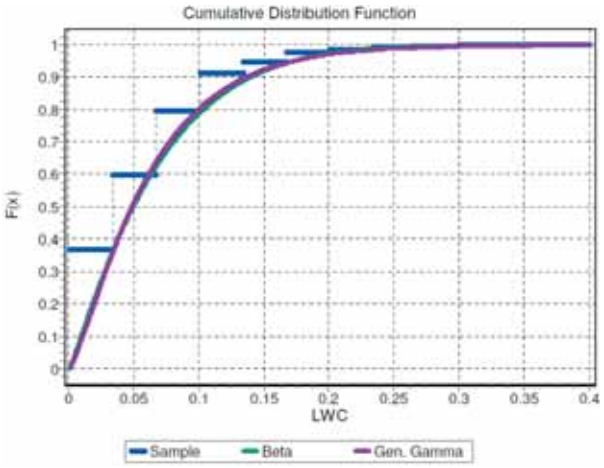

Fig. 7 CDF of the measured LWC

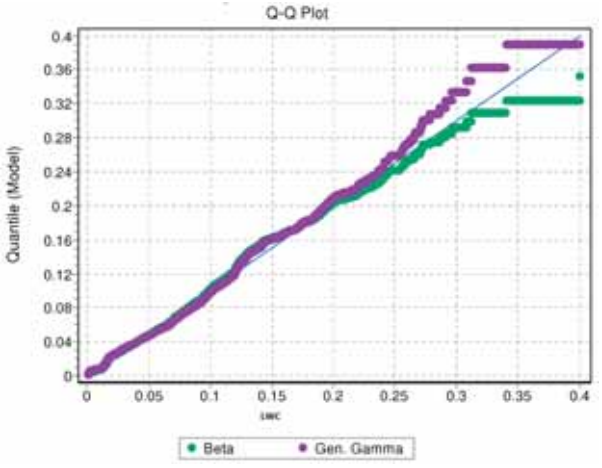

Fig. 8 QQ Plot for measured data of LWC

We performed the goodness of it by observing the QQ plot of the measured LWC. The QQ plot is provided in Fig. 8. It is obvious from Fig. 8 that Gamma distribution model is performing well as compared to Beta distribution. QQ plot for Gamma distribution is converging but QQ plot for Beta distribution is deviating for higher values of LWC. The distribution function for three parameters Gamma distribution is provided in equation (2)

$$f(x) = \frac{(x-y)^{\alpha-1}}{\beta^{\alpha}\Gamma(a)}\exp\left(-(x-y)/\beta\right) \qquad (2)$$

Where α is a shape parameter ($\alpha > 0$), β is scale parameter ($\beta > 0$) and γ is location parameter. We computed the optimum parameters for Gamma distribution for measured visibility which are provided in Table 2.

We concluded that Gamma distribution (3 parameters Gamma Distribution) is the best suited model for measured LWC in the absence of precipitation. The proposed distribution model can be used under fog conditions in the absence of precipitation.

4. Conclusions

A detailed statistical analysis of 5 months real time measured data of visibility and LWC for terrestrial FSO links has been presented to find the best fit distribution model for visibility and LWC. Our findings show that in reduced visibility, due to fog, 4 parameters Gamma distribution best fit the measured data of visibility and 3 parameters Gamma distribution is the best fit distribution model for LWC in the absence of precipitation. Both proposed model can be used under fog conditions ranging from dense fog to light fog and in the absence of precipitation.

Acknowledgments

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Lourdes Beloqui Yuste – Hugh Melvin *

A PROTOCOL REVIEW FOR IPTV AND WEBTV MULTIMEDIA DELIVERY SYSTEMS

This paper reviews the key protocols used for multimedia delivery both over privately owned managed IP Networks such as IPTV and public non-managed IP Networks such as WebTV. Within these two worlds, the choice between protocols is based on the characteristics of the multimedia service required, the media server and the client's receiver.

Users have different expectations of these two delivery platforms and also of different output devices such as TV, PC, tablet or mobile phone. In an IPTV environment, which is a paid-service, users demand the Quality of Service (QoS) they pay for; different quotas provide different services to users. On the other hand, with WebTV, which is a free service, clients have lower quality expectations.

There are multiple multimedia delivery systems used across both platforms, using different protocols to deliver multimedia from server to one or multiple clients. These include Real-Time Protocol (RTP), Real-Time Control Protocol (RTCP), Real-Time Streaming Protocol (RTSP), Real-Time Messaging Protocol (RTMP), Hypertext Transfer Protocol (HTTP), HTTP Live Streaming (HLS) and Microsoft Smooth Streaming Protocol (MS- SSTR).

RTP, RTCP and RTSP are the main protocols used in IPTV while RTMP, HTTP, HLS and MS-SSTR are protocols used mainly for WebTV. This paper will explore the main differences and similarities between them and the reasons behind the choice of one or the other. This review paper also briefly outlines a testbed being developed by the authors to synchronise media streams using a subset of these protocols.

Keyword: *Index Terms–IPTV, WebTV, Media Streaming, RTP/RTCP, RTSP, HTTP, MS-SSTR, RTMP*

1. Introduction

Multimedia traffic over IP Networks includes services such as Internet Video, Voice over IP (VoIP), IPTV, WebTV and Video Calling [1]. Such data transmission requires specialised protocols to accomplish the user quality requirements which for some services includes hard real-time delivery.

The growth of IP Networks for media delivery is increasing day by day. Cisco White Paper [1] and [2] provided interesting data that charts the increase IP Network use and the increase of Internet traffic dedicated to Video. Fig. 1 depicts the 2010 broadband traffic by application subcategory [1] and Fig. 2 shows the global consumer video 2010-2015 by category per month [2]. According to [2], projected data growth over IP Networks between 2010 and 2015 will be 55%, comprising a 32% increase of fixed Internet, and 24% of Managed IP and 92% mobile Internet. Internet video traffic in 2010 was 40% of total traffic although Cisco also forecast that this will reach 62% by 2015. Even geographical areas traditionally behind in Internet Video such as The Middle East and Africa will grow by 105%. Interesting data published in [1] details how Internet Video tripled in 2010 and will grow 17-times by 2015.

In 2010 Internet Video watched on a TV set traffic was 7% of Internet video traffic, this is projected to increase to 16% by 2015 [2]. The same source predicts that Video-on-Demand (VoD) will triple by 2015. In 2011 VoD High Definition (HD) video will surpass Standard Definition (SD) and it is projected that 77% of VoD traffic will be HD by 2015.

The growth of IP Network traffic and the increase in traffic dedicated to multimedia applications (both hard real-time and soft real-time) has led to major developments in protocols and technologies used for multimedia delivery. Multimedia transmission has differing characteristics to other data such as email, etc., such as high network load, real-time delivery and other quality performance criteria.

The delivery of multimedia over IP Networks is typically analysed by two parameters, QoS and Quality of Experience (QoE). The former is based in objective network analysis using metrics, such as delay, jitter and packet loss, whereas the latter is based on the user's subjective opinion about the final media play-out.

Real-time delivery is the main characteristic which distinguishes media delivery from others. Many Internet applications require lossless delivery of information via Internet, such as email where users want to receive the exact content sent by the sender. Multimedia delivery on the other hand is more tolerant of loss but

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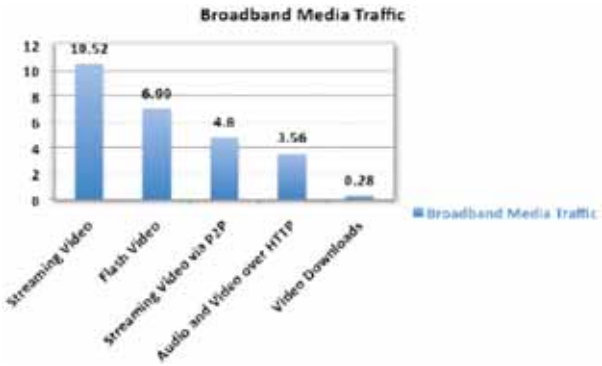

Fig. 1 2010 Broadband Traffic by Application Subcategory. A total of 26.15% dedicated to Online video [1]

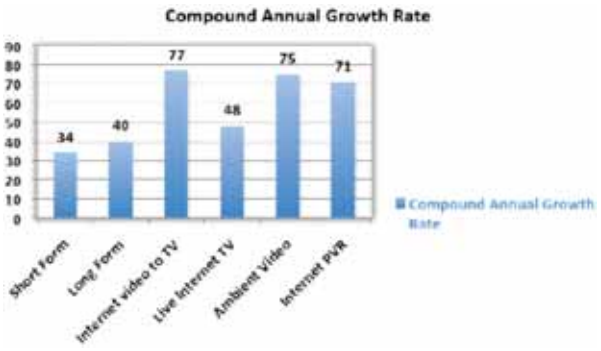

Fig. 2 The Global Consumer Internet Video 2010-2015 by Category per month [2]

WEBTV and IPTV differences Tab. 1

Symbol	webTV	IPTV
Hardware	PC	TV and STB
Software	Browser-based HTTP media selection	Media player Electronic Program Guide (EPG) channel selection
Network	Multiple protocols Public Unmanaged Worldwide Mainly unicast Best effort quality	RTP/UDP Private Managed Local Mainly multicast QoS guaranteed
Media	Unprotected Multiple coding Access to all Internet media Free media	Protected by encryption SDTV/HDTV Limited to IPTV content Paid provider media
User	High level involvement	Low level involvement
	Unsafe, unknown users Free access Free service	Safe, authenticated users Access only to known users Monthly payment

requires that data is received within a time range; otherwise it is not valid. For example, for VoIP and video conferencing, M2E (Mouth to Ear) delays typically should not exceed a round trip time delay of 300msec. Often, relative arrival delays, rather absolute delays, are more important e.g. when we are watching a video over the Internet if the sound related to the image we are watching arrives much later it does not make sense to play it and thus it should be ignored. On the other hand it is better to receive an image of lower quality than one out of synch with audio.

There are two main factors that determine which protocols are used, the media platform and the Transport Protocol used underneath the Application Protocol.

Within the TCP/IP protocols stack, all layers have a big roll to play. Application Layer (AL) protocols are highly dependent on the Network and Transport Layer (NL/AL) protocol used. In this paper we examined protocols mostly at Application and Transport layers and how they are related. Media traffic can use IP unicast or multicast at the Network Layer and User Datagram Protocol (UDP) and Transmission Control Protocol (TCP) at the Transport Layer.

There are two main IP Network delivery platforms, IPTV and WebTV. Due to their different characteristics each one uses different application layer protocols. While IPTV uses RTP combined with RTSP, WebTV is mainly HTTP oriented. HTTP is not a media delivery protocol but the main protocols used in WebTV are based on HTTP media delivery.

The remainder of this paper is structured as follows. Section II depicts the media delivery platform, Section III describes the different media delivery systems such as downloading, progressive downloading, streaming and adaptive downloading. Section IV presents the IP, Transport and Application protocols related to media delivery. Section V briefly describes the testbed developed, and finally Section VI concludes the paper.

2. Media Platform

The media distribution over IP Networks can be performed via two different systems, WebTV and IPTV. Although both use IP they have different characteristics that affect the media delivery protocol used. The main differences are established by Maisonneave in [3] and they are depicted in Table I. An important fact from the user's perspective is their expectations based on cost. Users paying a monthly quota for e.g. for IPTV expect a high QoS and QoE and they expect to receive the services they pay for. Users using free WebTV will accept poorer quality media delivery.

The other aspect is the protection of the media delivery which guarantees media providers that the media will not be illegally

TCP and UDP differences Tab 2

TCP [4]	UDP [5]
Connection Oriented	Connectionless
Guaranteed delivery of data	No guaranteed delivery of data
Flow control provided	No flow control provided
Error mechanisms provided	Basic error mechanism via check sum
Lost packets retransmission	No lost packets retransmission
Eliminated packet loss	Reduces packet delay

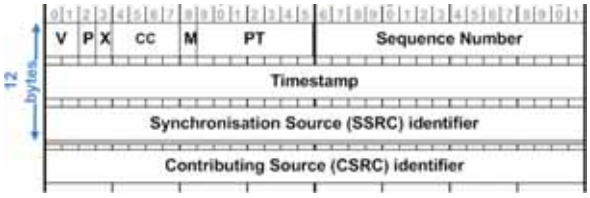

Fig. 3 RTP Header [13]

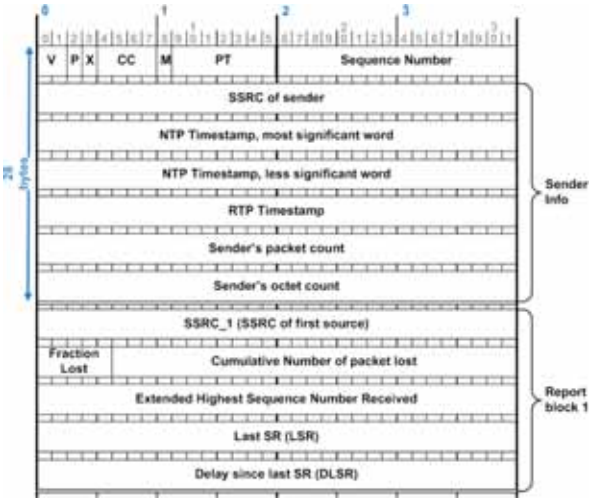

Fig. 4.a. Sender Report (SR) RTCP Packet Header [13]

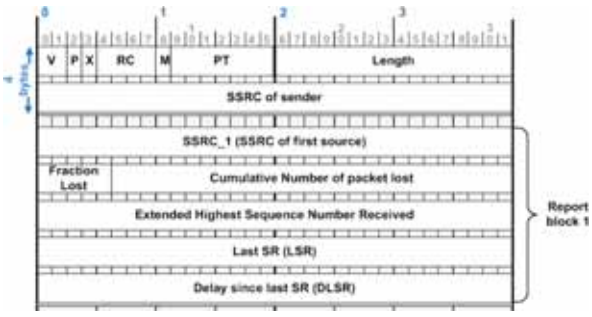

Fig. 4.b. Receiver Report (RR) RTCP Packet Header [13]

copied or distributed. Without network protection IPTV companies would never be able to buy the rights to distribute the media content which is the core of their business.

Both systems use IP delivery but the IP protocol used in each case is different. IPTV uses IP Multicast for video delivery, only employing IP unicast for VoD whereas WebTV mainly employs IP unicast protocols.

3. Media Delivery

There are a few different techniques to deliver and play a media file delivered from the source to the receiver. Different techniques range from downloading and progressive downloading to adaptive streaming and streaming. Downloading and streaming are the most extreme methods whereas progressive downloading and adaptive streaming are hybrid techniques which keep the advantages of downloading and streaming techniques while striving to avoid their drawbacks.

The transport protocol used in the media delivery is a key factor also. The most common transport protocols are TCP [4] and UDP [5]. TCP provides retransmission of lost-packets to guarantee integrity. However it does not assure on-time delivery, whereas UDP provides a low-overhead protocol to provide, quicker, albeit best-effort delivery with no loss-packet retransmission. Table II provides the main differences between both transport protocols revealing the main functionality of UDP to reduce packet delay and traffic overheat to benefit Real-Time delivery.

UDP is typically used by RTP/RTCP, described in Section IV.A, for streaming media delivery. This contrasts with HTTP based protocols, such as HLS and MS-SSTR explained later, which use TCP for downloading, progressive downloading and adaptive streaming methods. Although RTMP uses TCP at Transport Layer it does not employ HTTP for the media delivery.

IPTV systems use streaming to deliver TV multicast services, although they can provide unicast VoD services via HTTP, whereas WebTV usually employs unicast progressive and adaptive HTTP streaming.

A. Downloading

The downloading method requires the download of the media file to the receiver prior to its display. The advantage is that once the file is downloaded at client-side the play-out is not interrupted. On the other hand the biggest drawback is the waiting time for the media file to be completely downloaded. The most popular protocol to download files is HTTP based in TCP. Its main drawback is the latency until the video play-out begins. As such it is mainly used for small videos with low quality where latency is minimised [5].

B. Progressive Downloading

HTTP Progressive download was developed to reduce the latency of initial play-out, while still preserving all the advantages

of HTTP over TCP. HTTP Progressive Downloading is an optimised HTTP technique to stream media over TCP. Media play-out begins before the media file is completely downloaded to minimise the user's waiting time before being able to start playing the video. Multiple web media providers such as YouTube use progressive downloading [6].

RTSP State and methods Tab. 3

State	Message received	Next state
Init: Server initial state and client waiting setup reply		
Init	Setup	Ready
	Teardown	Init
Ready: Server setup, play, pause sent		
Ready	Play	Private
	Setup	Managed
	Teardown	Local
	Record	Recording
Playing: Client has received a play reply from server		
Playing	Play	Playing
	Pause	Ready
	Teardown	Init
	Setup	Playing
Recording: Server recording and client received record reply		
Recording	Record	Recording
	Pause	Ready
	Teardown	Init
	Setup	Recording

C. Streaming

The principal objective of streaming is the media delivery at real-time when the main purpose is the media delivery within a time threshold. The underlying protocol used is UDP because it provides a low overhead and minimises packet delay delivery. On the other hand this is also its main drawback as UDP is often blocked by firewalls and penalised by Internet providers.

Streaming benefits include minimising the buffer size at client, low latency, no need to store media file at client-side, efficient use of the bandwidth, and providing the means to analyse the media selection user's behaviour, thus making it the most appropriate delivery method for live video [6] [7].

D. Adaptive Streaming

HTTP adaptive streaming uses progressive downloading to deliver the same multimedia content adapted to each individual client. Once the media session has been initialized, on client request or based on network conditions, media can be adapted, for example, to different bit-rate or quality [7].

4. IP/Transport/Application Protocols

The Application protocol used for real-time media delivery depends on a range of factors, including the media platform, IPTV or WebTV, IP Protocols, IP unicast or multicast, and the Transport protocol, TCP or UDP. In IPTV the main protocol is RTP over UDP using IP multicast whereas WebTV utilizes HTTP based protocols using IP unicast.

IP multicast is widely used for IPTV service providers to deliver channels to large numbers of clients with IP unicast used for their VoD services. WebTV, although it can utilize IP multicast, is mainly delivered over IP unicast.

HTTP is used by many companies to develop their own HTTP based Application protocol. Apple is currently finalising the Internet-Draft HLS and Microsoft has created their own MS-SSTR. All of these protocols use HTTP progressive download techniques although they are clearly differentiated. Adobe Flash has developed RTMP which is also based on TCP but not based on HTTP.

Typically the relationship between protocols is dictated by their functions. RTP provides functionality similar to HTTP, both actually delivering the media data, and RTSP relates to RTMP, HLS and MS-SSTR because they organise the complete media session and manage information about the network conditions, the media, the server and the client.

A. RTP/RTCP/RTSP

The benefits of media transport using RTP have been widely studied; [8], [9] and [10] focused on MPEG-2 media streams whereas in [11] and [12] the benefits of the use of RTP on MPEG-4 media streams are explained.

RTSP and RTP/RTCP are mainly used in IPTV systems. RTSP in conjunction with RTP, and its companion RTCP, provide a protocol suite for real-time data delivery such as video and audio. RTP

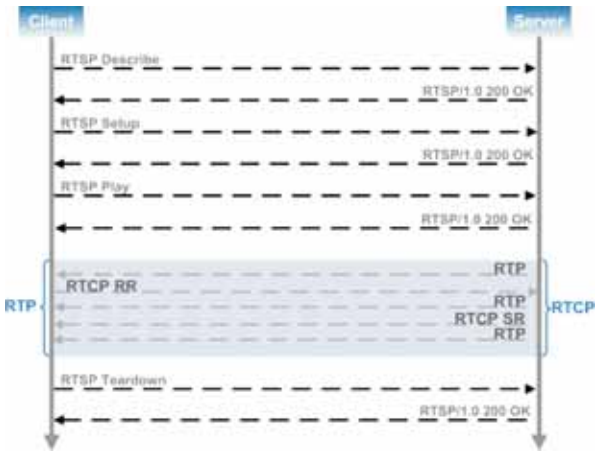

Fig. 5 RTSP Communication [14]

transports the media data, audio and video, while RTCP monitors the media delivery and RTSP controls the delivery of the real-time data.

RTP header provides three important fields for the real- time media delivery which are *timestamp, sequence number and payload type* (PT). The first has a different meaning depending of the payload type, the second helps to reorder the packets at client-side due to the fact that RTP, via UDP, does not provide ordered transport delivery. Finally, the *payload type* is also important as it indicates the media type conveyed within the RTP packet. In Fig. 3 the RTP packet header is depicted.

Once the RTP transport of real-time media streams is initiated, RTCP packets are sent between media session senders and receivers to monitor the media delivery. The media sender sends RTCP Sender Report (RTCP SR) packets and receivers send RTCP Receiver Report (RTCP RR) packets. Those RTCP report packets are different. In Fig. 4a the RTCP SR Packet Header shows important fields such as *NTP timestamp, RTP timestamp* whereas Fig. 4b shows the RTCP RR important fields such as *accumulative number of packet lost, last SR and delay since last SR* [13].

Fields such as *inter-arrival jitter, delay since last SR* and *cumulative number of packet lost* provide information to monitor the media transmission within a media session. The *inter-arrival jitter* provides the inter-arrival time variance of the RTP data packets, *delay since last SR* indicates the time in seconds between the reception of the last two RTCP sender packets, and finally *cumulative number of packet lost* conveys the quantity of packets lost from a source since the beginning of the media session [13].

RTSP maintains the state of the media session. Furthermore with the states and methods, RSTP provides the tools to provide remote control functions over IP Networks. The state is needed to relate a media stream to an RTSP request. States for both client and server are Init, Ready, Playing, and Recording. Not all methods provoke a change of state, *options, announce, describe, get-parameter,* and *set-parameter,* only provide information about the media session whereas *setup, play, pause, teardown, redirect,* and *record* invoke a change of state. Table III depicts the change of states and a description of the client and server states [14].

Fig. 5 depicts the RTSP communication process previous to the RTP media data transport, which includes RTCP packets interchange between client and server, and the media session finalisation. It is important to note that while RTP is transmitted using UDP, RTSP packets are usually sent using TCP to mitigate packet loss.

The process sends an RSTP *Describe* method to the server to require information about the media session. The server responds sending the relevant information via Session Description Protocol (SDP) [15], which is outside the scope of this paper. The client receives media information and sends to the server an RTSP *Setup* command which is responded by server with an RTSP *OK* message. After the initiation of the media session is finalised the client sends an RTSP *Play* and the server finally responds and begins the media transmission via RTP packets. During RTP transmission RTCP SS and RS packets are sent by the sender and receiver respectively. When the client desires to finish the media session sends an RTCP *Teardown* command to the server which accepts with an RTSP *OK* response.

B. RTMP

Adobe Flash Platform technology utilizes RTMP to stream audio, video and data over TCP. This method is designed to stream flash encapsulated media between server and client. It is used in the Internet to transmit both live Internet Radio and WebTV.

As an introduction to RTMP a few concepts such as message stream, chunk and chunk stream need to be clearly specified.

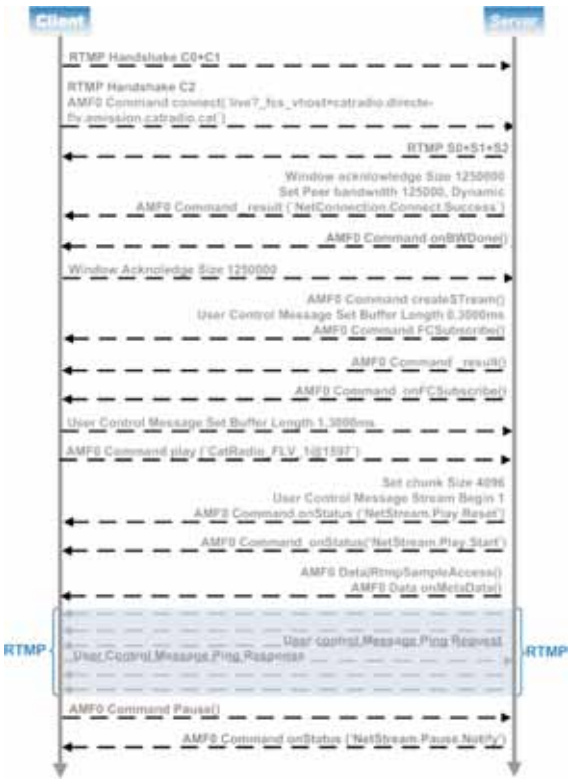

Fig. 6 RTMP Communication connection to http//www.catradio.cat/endirecte

The message stream is defined in [16] as 'A logical channel of communication that allows the flows of messages'. Messages are fragmented and interleaved to send over the network. A message stream can be an audio, video or data stream.

A chunk is a message fragment that facilitates timestamp ordered delivery, from server to client, of the complete message. Also in [16], a chunk stream is defined as 'A logical channel of communication that allows flow of chunks in a particular direction'.

HTTP versus RTP streaming Tab. 4

HTTP streaming	RTP streaming
Firewall friendly	Firewalls block UDP traffic
HTTP traffic allowed by Network providers	Network providers block or penalize UDP traffic
HTTP-based web server	Specialised media streamer server
Pseudo real-time delivery	Real-time streaming
Pull-based	Push-based
Lack rate control	Rate control
Content stored at client	Content buffered at client
No efficient bandwidth use	Efficient bandwidth use
Longer latency	Minimal latency
Guaranteed content delivery	Unreliable content delivery
Minimises packet loss	Minimises packet delay

Chunks are parts of the audio and video streams, called audio/video messages streams, delivered to client and reconstructed into messages. Small messages can be sent within a unique chunk, while a message's partition into chunks is required when messages are bigger than the maximum allowed chunk size within the message.

RTMP specification consists of the RTMP Chunk Stream Protocol and the RTMP Message Protocol. The former details the message format to convey message chunks and the initial handshake to establish the multimedia connection/session, whereas the latter describes the RTMP message formats, RTMP control and command messages.

Two main concepts are part of the RTMP, NetConnection and NetStream. The former creates a client-server connection whereas the latter represents the communication channel used to deliver audio, video and data streams. Commands are applied to both in a media session. NetConnection commands are *connect, call, close,* and *createStream* while NetStream commands are *play, play2, deleteStream, closeStream, receiveAudio, receiveVideo, publish, seek,* and *pause*.

RTMP first of all establishes a NetConnection via a handshake, initiating interchange of packages between client (packets C0, C1 and C2) and server (S0, S1, S2). Secondly the server provides the client with information about the media session such as Window acknowledge size, bandwidth, set chunk size and buffer length. The final step is to play the NetStream indicated in the transaction.

The message header has four fields *Message Type* (1 byte), *Length* (3 bytes), *Timestamp* (4 bytes) and *Message Stream ID* (3 bytes). This message header is used to transmit control and command messages as well as to initialise a stream message. The chunk header is more complex, being composed of four fields *Basic Header, Chunk Message Header, Extended Timestamp*. The complexity comes from three types of Basic Header plus four types of *Chunk Message Header*.

In Fig. 6 the communication workflow to access a live Internet Radio media is depicted. First the RTMP handshake between client and server is performed via transmission of C0, C1, C2, S0, S1, and S2. Then information about the netConnection and netStream is interchanged and finally the media data transmission is performed via RTMP chunks. Finally the connection client sends a pause command to the server which acknowledges the message.

C. HTTP

HTTP was initially conceived and designed to deliver static documents and consequently rate control was never provided and the underlying use of TCP, with retransmission of lost packets, causes variation in media delivery [17].

HTTP streaming has numerous advantages over RTP streaming mainly caused by the protocol used at the transport layer, TCP. The main differences from [6], [7] and [17] are depicted in Table IV which explains the relative popularity of media delivery via HTTP streaming, particularly for WebTV.

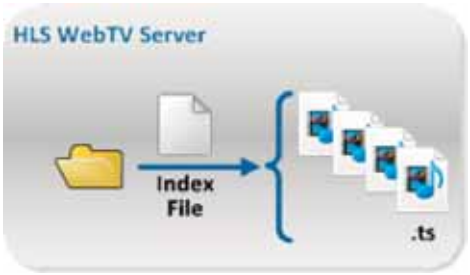

Fig. 7 HLS webTV Server file organization. The index file indicates where are located the .ts files storing fragments [22]

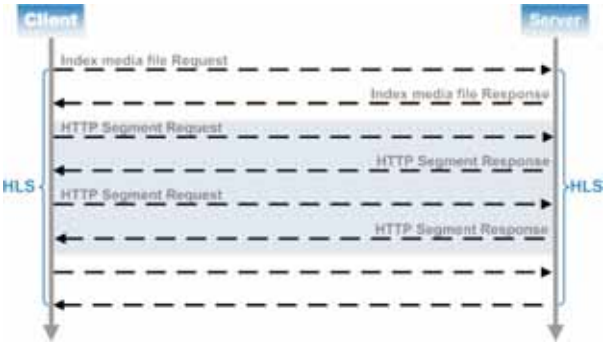

Fig. 8 HLS Communication [22]

Fig. 9. High level ISO file format structure with multiple fragments used by MS-SSTR [19]

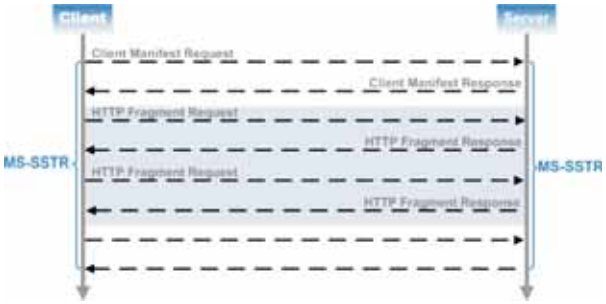

Fig. 10 MS-SSTR Communication [19]

HTTP mainly uses TCP as a transport protocol and is widely used in WebTV. It is a stateless application layer protocol, based on a connection between client and server to provide a communication channel. The basic communication unit is a message; the client sends to the server a HTTP request message and the server answers with the HTTP response message. HTTP also provides a list of methods applied to the message sent. It is important to note the possibility to transfer a message divided into chunks when needed [18].

One main difference between HTTP streaming and other HTTP transactions is that it does not terminate the response to the client by the server due to the constant request of media segments by the client. HTTP streaming uses persistent connection, *Connection: Keep-Alive*, to receive all media segment client requests.

HTTP based progressive download has been further developed into the latest HTTP adaptive streaming. There are multiple options among which are HLS developed by Apple and MS-SSTR by Microsoft. The differences between HTTP based protocols HLS and MS-SSTR are depicted in Table V.

1) HTTP Progressive Download: HTTP progressive download is a hybrid technique between HTTP download and RTP streaming. It downloads the media file in very small parts instead of downloading the entire media file before the client's play-out. To accomplish that, the server stores the media file in small segments, usually from 2sec to 20sec, and an index file with segment transmission, segment information and media resources. The process begins by sending the requested media resource index file to the client. Then streaming via HTTP begins with an HTTP request for the segments and the download. [7] An example of HTTP Progressive Download is HLS provided by Apple and described in subsequent section IV- D.

2) HTTP Adaptive Streaming: Adaptive streaming utilizes HTTP progressive download to accomplish streaming features with added functions such as adapting the media delivery to the client's end device or to Network conditions. Adaptive streaming exploits one of the HTTP characteristics which is to deliver a video/audio file fragmented into small segments or chunks. The media file is stored at the server in multiple small segments or in a unique file virtually fragmented into segments [19].

Segments or chunks are usually 2-4 seconds long and usually begin with an I-frame as well as following GOP boundaries to provide chunk independence from previous or posterior chunks to further facilitate the separated chunk decoding at client-side [19].

A server can store the different versions of the same media file and deliver to a client one version or the other. The same media file can be stored in different quality levels or different bit-rates. The segment stored system allows clients to change the version delivered during streaming, reacting to a client's petitions, the client's end device or to varying network conditions [7].

MS-SSTR and HLS differences Tab. 5

HLS	MS-SSTR
Apple	Microsoft
Uses with a HTTP server	Needs a server-specific IIS extension
One Index file	Client and Server Manifest file
Index files use M3U8 specification	Client's manifest use XML format
H.264 visual codec	H.264 visual codec
MP3 and AAC audio codec	AAC audio codec
Each segment stored in a ts file	One MP4 virtual fragmented file
Media divided into media segment	Media divided into fragments

Protocols characteristics Tab. 6

RTSP	HLS	MS-SSTR	RTMP
IPTV	webTV	webTV	webTV
RTP packets	HTTP segments	HTTP fragments	RTMP chucks
IETF	Apple (IETF)	Microsoft	Adobe Flash
TCP	TCP	TCP	TCP
MP2T	MP2T	MPEG4 part 14	Multiple
Stateful	Stateless	Stateless	Stateless
No handshake	No handshake	No handshake	Handshake connection

Research in adaptive streaming includes the delivery of real-time content such as sports [20]. The latest research in media delivery at mobile platforms uses adaptive streaming to adapt the media delivery to a device's battery consumption.

D. HLS

HLS is an IETF open standard Internet-Draft developed by Apple. Only the latest versions of QuickTime on IPhone 3.0 support the protocol although any HTTP server can use it to deliver the media data. [21] It follows all characteristics of HTTP Progressive Download, i.e., media delivery via HTTP of small fragments, chunks, and the initial media play-out before the complete media download. Finally the HTTP server stores the media source in different bit-rates and one or the other is chosen depending on client or network conditions. In Fig. 7 the HLS server file organization

is depicted and the HLS communication process between server and client is described in Fig. 8.

The media container is restricted to MPEG-2 Transport (MP2T) streams. Although in theory there are no media codec restrictions, currently only AAC and MP3 for audio and H.264 for video are accepted. Files are stored at server-side in MP2T files. A single media stream is stored in multiple MP2T files, one for each fragment called media segment files. [22] The index file, also located at server stores the information to locate the MP2T media segment files required to play the entire media stream [21].

The index file follows M3U8 specification which is a play list specification extended from MP3 play list (M3U). [21] Every media is encoded in different bit-rates and has its own index file, as shown in Fig. 7. Index files contain metadata with information about the sequence in which to play the fragments related to a single media source, the location of the MP2T files containing the segments and the alternative media files available in case the request bit-rate is not available [21].

The client requests the media source index file from the server. As a result the server sends it to the client, as shown in Fig. 8. The client in possession of the index file is able to request the fragments in the right order, indicating their server location for the media stream play-out [21].

E. MS-SSTR

MS-SSTR is based on HTTP streaming and designed by Microsoft. It uses adaptive streaming to adapt the media delivery based on local bandwidth and client's CPU. Its main characteristics are its exclusive use of MP4 container, 14496-14 [23], and the use of two index files. First of all the media file uses MP4 format and is structured by splitting the media into smaller fragments within the file and secondly it uses two index files, the server and client manifest [7].

Both manifest index files are stored at server. The server index manifest file provides the server information about the media tracks, bit rates and files, whereas the client index manifest lists the tracks available to the client [7]. In Fig. 9 the ISO file format structure used by MS-SSTR is depicted.

The communication model is based in manifest and fragment request. Assuming an HTTP connection between client and server, the client's first step is to request the Client index manifest which is delivered by the server via the manifest response. After the positive reception of the manifest file, client begins the fragment request to server which sends back the fragment response. This process, fragment request and response, is repeated multiple times until the completed media delivery is terminated [24]. In Fig. 10 the HLS communication process between server and client is described.

The manifest index file is used by the client to request the media fragments based on the timing and bit-rate information. The server, considering network conditions, relates the fragment's request to the corresponding MP4 file where the fragment is extracted and sent as an independent file [7].

One of the main advantages of MPS-SSTR is the use of MP4 file format. The use of this media container facilitates smooth streaming. The media file is stored in an MP4 format which stores all fragments containing the media chunks to deliver in each fragment request.

5. Testbed Development

This protocol review provides a foundation for our experimental research work. We are developing a media synchronisation testbed to evaluate different synchronisation scenarios using media delivered by different methods. A related concept is Hybrid Broadband Broadcast TV (HbbTV) which provides users with IPTV, WebTV and Broadcast TV on a single device. One scenario being evaluated is to synchronise a TV channel delivered via IPTV and a Radio channel delivered via Internet. The TV channel is delivered using RTP and MP2T as indicated in [25]. Regarding the Internet Radio channel, the delivery method can vary. The first step has been to stream to the client via RTP over MP3 and synchronise this mp3 audio with the related video stream. In future work, other options deploying other delivery methods such as HTTP or RTMP will be deployed.

The integration of video and radio streams using RTP provides us with the simplest media delivery method. So far two approaches have been coded, audio substitution and audio addition. The audio substitution replaces the audio from the TV channel and the audio addition creates a new audio channel with the audio from the Internet Radio allowing the user to switch from the original audio and the new audio and vice versa.

We anticipate that such services will be of most potential in the live sporting domain. The media files used in our testbed are related to the same sport event, the Champions League Final FC Barcelona vs. Manchester United year 2011. The video is from SkySportsHD channel with English audio, in MP2T format. The Radio media file, in MP3 format, is the radio transmission of the same event from the Catalan National Radio Station Catalunya Radio.

6. Conclusion

Media traffic over IP Networks is constantly increasing and various protocols are used, each of them suited to particular scenarios. The choice of protocols depends on a range of factors, including, the media platform, IPTV or WebTV, the Transport protocol used, TCP or UDP, IP protocol multicast or unicast, and the server providers' technology.

In this paper we have reviewed the overall suite of media protocols, outlining how they work both individually and in tandem with others to deliver media streams. IPTV systems typically use

RTP and RTSP protocols over UDP and the latest DVB-IPTV standards confirm the future use of these application protocols. Meanwhile WebTV preferences are moving towards HTTP over TCP as a media delivery. The main concern is the multiple HTTP server providers each of which uses their own technology. The principal differences between the protocols are listed in Table VI.

A further complication arises as multiple vendors have developed their own protocols. RTMP, by Adobe Acrobat, HLS by Apple, and MS-SSTR by Microsoft are the main ones. We also briefly outlined our research testbed, which focuses on multimedia synchronisation challenges and opportunities.

Acknowledgement

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Tatiana Kovacikova – Pavol Segec *

TESTING FRAMEWORK FOR IMS-BASED IPTV

IPTV (Internet Protocol television) is one of the fast-growing communication services of the past years. Unfortunately, there is a lack of standardized deployed solutions and operators need to to rethink their position before upgrading IPTV solutions for next generation TV (HD TV and 3D TV). An efficient and up-to-date Interoperability Test Specification for IMS-Based IPTV is one of the opportunities for service providers and network operators to evaluate different features and services. TISPAN Release 3 covers many new use cases and features for IPTV (like user generated content, target advertising, etc.) and possible interactions with Voice/Data communications such as Social TV, Incoming Voice call management and notification on TV screen. In this paper, we will first provide a general overview of interoperability testing and its features. On an Instant Messaging use case, we will provide test descriptions including an end user test sequence as well as high level message flows at key standardized reference points in the TISPAN IMS-based IPTV infrastructure.

Key words: *IPTV (Internet Protocol Television), interoperability, testing*

1. Introduction

The current and future e-Communications market can be seen as a convergent multimedia market with an increasingly complex structure. Within this market we are faced with an unpredictable, sometimes fragmented, market development where potential barriers to achieve interoperability may be emerging. Additionally, within the present competitive environment, the risk of non-interoperability is increasing because of fast evolution of technology provided by different vendors, or the use of non-open standards.

Against this background there is an ever-increasing awareness of market players and regulators that mass-market development requires interoperability based on open standards. Additionally, the end-user appreciates more choice, but expects certainties.

Interoperability between devices, applications, data repositories, services and networks is a key requisite for taking advantage of the benefits of digital technologies. New standards are being created all the time, globally, and Europe's standard-setting framework must catch up with fast-moving technology markets if it is to remain relevant for the future. The main aim of standardization is to enable interoperability in a multi-vendor, multinetwork, multiservice environment. The absence of interoperability must not be the reason why final services for which there is great demand do not come into being.

With the introduction of IPTV, traditional wireline service providers and content providers are entering a new market delivering broadcast and VoD (Video on Demand) services to consumers. The success of this endeavor is heavily dependent upon how fast service providers can roll out reliable IPTV services that give consumers the most convenience and flexibility before competing providers do the same thing. Unfortunately, most of the deployed solutions are not standardized and operators will have to rethink soon their position before upgrading IPTV solutions for next generation TV (HD TV and 3D TV). An efficient and up-to-date Interoperability Test Specification for IMS-Based IPTV is one of the opportunities for service providers and network operators to evaluate different features and services. ETSI TISPAN Release 3 covers many new use cases and features for IPTV (like user generated content, target advertising, etc.) and possible interactions with Voice/Data communications such as Social TV, Incoming Voice call management and notification on TV screen. In this paper, we first provide a general overview of interoperability and its features. Then, we describe the ETSI TISPAN approach to IPTV. For one of them, the IMS-based IPTV, the testing framework is illustrated. Finally, on an Instant Messaging use case, we show test descriptions including an end user test sequence as well as high level message flows at key standardized reference points.

2. Different Levels of Interoperability

There is a number of definitions of the word interoperability: even at ETSI (European Telecommunications Standardization Institute) the term has different meanings in different contexts [1]. However, the following definitions are probably the closest to our understanding:

- "Interoperability is the ability of equipment from different manufacturers (or different systems) to communicate together on the same infrastructure (same system), or on another while roaming" from ETSI's Technical Committee TISPAN.

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Interoperability can be seen at different levels: technical interoperability, syntactical interoperability, semantic interoperability and organizational interoperability.

Technical interoperability is usually associated with hardware/software components, systems and platforms that enable machine-to-machine communication to take place. This kind of interoperability is often centred on communication protocols and the infrastructure needed for those protocols to operate. Syntactical interoperability is usually associated with data formats, as messages transferred by communication protocols need to have a well-defined syntax and encoding, either in the form of bit-tables, or they can be represented using high-level transfer syntaxes such as HTML (Hypertext Markup Language), XML (eXtensible Markup Language) or ASN.1 (Abstract Syntax Notation). Semantic interoperability is usually associated with the meaning of content and concerns the human rather than machine interpretation of the content. Organizational interoperability is the ability of organizations to effectively communicate and transfer data and information even though they may be using a variety of different information systems over widely different infrastructures, possibly across different geographic regions and cultures. It can be said that organizational interoperability is the high-level issue and in general, it depends on successful technical, syntactical and semantic interoperability.

The development of standardized test specifications is an integral part of the ETSI strategy for ensuring interoperability. In the context of standardization ETSI focuses on the development of two types of test specifications, which reflect the principle: test the components first, then test the system, i.e.:

- Conformance test specifications; and
- Interoperability test specifications.

Conformance testing concentrates on specific components in a system, often related to a single standard (or a set of related standards). It is unit testing rather than system testing. Conformance testing is applied over open interfaces and checks for conformance to the requirements in a base specification or standard. Interoperability testing concentrates on a complete device or a collection of devices. It is system testing rather than unit testing. It is most commonly applied to end-to-end testing over networks. It shows, from the user's viewpoint, that functionality is accomplished, however, it does not show how.

Both, conformance as well as interoperability testing are important for successful deployment of the equipment from different vendors and its interworking [2]. The both should be regarded as complementary techniques. Many certification schemes require, for example, conformance testing as a prerequisite to interoperability testing (e.g. the Open Mobile Alliance or the WiMax Forum).

3. TISPAN approach to IPTV

While all major service providers are planning to launch or have already launched IPTV services, the strategies that will lead to the long-term economic success of these services are still being debated. A complex architecture is required to deliver competitive services, requiring close interoperability of all components involved in service delivery, including the Business Support System (BSS) and Operations Support System (OSS) and particularly the underlying network from the headend and VoD server to the home environment [3]. Within the home, there are other considerations in order to allow interworking of the STB with the control software (or middleware) and the integration of the middleware with other components (such as the headend, VoD servers, Web portals, and Electronic Program Guide (EPG)).

Aside from offering IPTV services, service providers have begun migrating their traditional fixed and mobile voice and communication services to converged IP NGN networks. As circuit-switched technology is phased out, new VoIP and rich media communication services are being deployed in packet-switched environments through the use of SIP signalling. IP Multimedia Subsystem (IMS) is emerging as an effective, standards-based architecture defining SIP-based voice and rich media service delivery. Today IMS is being promoted as the architecture of choice for multimedia communications services of all kinds. The ETSI and CableLabs® are creating standards to enable an IMS architecture to be supported on their specific access networks. ETSI Technical Committee TISPAN is the standards group within ETSI that is developing a more universal service delivery architecture that adapts the 3GPP-defined IMS standards to address the needs of wireline providers to build a policy- controlled IP transport network. Current versions of the IMS standards are focused on SIP-based communication services and PSTN replacement. Standards are still a work in progress, with 3GPP at Release 11 and TISPAN working on the definition of an NGN Release 3.

As part of its NGN (Next Generation Networks) Release 2 series of specifications, ETSI TISPAN has defined the integration of IPTV services in an NGN architecture to answer the needs of network service providers and equipment vendors. TISPAN defines two solutions for the integration of IPTV in the NGN architecture: Integrated IPTV subsystem and IMS-Based IPTV.

The Integrated IPTV subsystem [4] focuses on the integration of existing market solutions in an NGN environment. Network service providers benefit from the cost advantages of a Next Generation Network without strong modification of their current IPTV service. For instance, the NGN integrated IPTV subsystem reuses all applicable DVB-IPTV (Digital Video Broadcasting – IPTV technologies).

The IMS-based IPTV solution [5] allows blending of TV services with other telecommunication services (e.g. voice, presence, and data services). Network services providers can take full benefit of the IP Multimedia Subsystem (IMS) architecture while providing key end-user services. In this paper, we focus on this solution.

The deployment of IMS functionalities to support IPTV services enables using a number of interesting IPTV features [6], for example:

- Integrated user registration and authentication (single sign-on);
- User subscription management;
- Session management, routing, service trigger, numbering;
- Interaction with existing NGN service enablers (presence, messaging, group management, etc.);
- Roaming and nomadism support;
- QoS and bearer control;
- Unified charging and billing.

The IMS based IPTV can also profit from additional features such as support for mobility, enabling interaction with existing NGN service enablers, service personalisation and media adaptation as well as to provide converged applications integrated voice, data, video and mobile services to flexible quadruple play service concept.

The high-level functional architecture for the IMS-based IPTV is shown in Fig. 1.

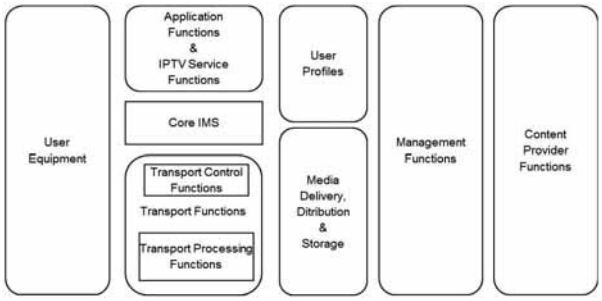

Fig. 1 High level functional architecture for the IMS-based IPTV

The IPTV enabled UE (User Equipment) terminates the IPTV control and media signals, and displays the corresponding information to the user. The UE interaction with the user allows selection of program, content, and service descriptions, such as content guides for broadcast and VoD services.

Application functions and IPTV service functions enable operation of or provide IPTV services. This includes IPTV Service Supporting Functions which represent common functions that could support or be used by other IPTV service or applications. Examples of IPTV service supporting functions may be Service Discovery and Selection functions.

User Profiles includes user data that are involved in providing IPTV services.

Core IMS provides functionality for authentication, authorization, and signalling for the setup of the service provisioning and content delivery. It routes signalling messages to the appropriate application server or triggers the applications based on settings maintained in the UPSF (User Profile Server Function). For resource reservation and admission control this function interacts with the RACS (Resource and Admission Control Subsystem).

Transport Functions include two kinds of functions:

- Transport Control: It contains functions from RACS and NASS (Network Attachment SubSystem). It provides policy control, resource reservation and admission control as well as IP address provisioning, network level user authentication and access network configuration as defined in TISPAN.
- Transport Processing Functions: They represent network access links and IP core. The IP core is in charge of data transmission with quality of service support.

The Media Delivery, Distribution and storage function receives and stores live feeds and media streams coming into the IPTV System from Content Providers. It is mainly in charge of media processing, delivery, storing, trans-coding and relaying. This function performs all these tasks along with the control of – or feedback to the IPTV Service and Control. Content protection may also be performed here or already protected content could be delivered over these functionalities.

The detailed specification of the architecture and functions of an IMS-based IPTV system is available in [5]. It should be noted that management functions and content provider functions are not specified in this document.

2. IMS-Based IPTV – Testing Framework

The IMS-based IPTV solution allows blending of TV services with other telecommunication services (e.g. voice, presence, and data services). Network services providers can take full benefit of the IMS architecture while providing key end-user services. Only a few basic use cases of IPTV have been defined in TISPAN Release 2. The IMS-based IPTV TISPAN Release 3 [7] covers many new use cases and features for IPTV (like user generated content, target advertising, etc.) and the possible interactions with Voice/Data communications such as Social TV, Incoming Voice call management and notification on TV screen. The IMS-Based IPTV Interoperability Test Specification for Release 2 is provided in the ETSI TS 186 020 Ver. 2.1.1 [8] published in 2009. The new test cases have been recently developed and linked to TISPAN Release 3 within the ETSI STF 413, one of the authors of this paper was a member of [9]. The updated testing specification was published as the ETSI TS 186 020 v.3.1.1 in July 2011 [10].

The specification covers the use of main IPTV functionality via different methods as defined in NGN Release 2 as well as NGN Release 3 new use cases and features for IPTV and possible interactions with Voice/Data communications such as Social TV, Incoming Voice call management and notification on TV screen. Interoperability test descriptions have been specified following the ETSI IPT test specification framework described in EG 202 568 [11] and interoperability testing methodology defined in EG 202 237 [12], i.e. interoperability testing with a conformance relation. Each interoperability test description includes an end user test sequence as well as a table for checking of high level message flows at key

standardized reference points in the TISPAN IMS-based IPTV infrastructure [5] and [7].

Fig. 2 shows the abstract test architecture of an IMS-based IPTV system based on the general IPTV architecture. In this figure, each node groups different IPTV logical functions. Interfaces within each node are considered internal and not taken into account in conformance criteria. Reference points (Ut, e2 and y2 towards BC-MCF) in dotted line are not in the scope of the specification. It should be noted that in a real IMS-based IPTV system some of the nodes shown in Fig. 2 may also be collocated in the same equipment. In this case it is however still assumed that their connecting interfaces are still available for monitoring purposes.

Each node framed with a solid line is considered Equipment under Test (EUT) in the context of the ETSI interoperability testing methodology [12]. The collection of all EUTs makes up the System Under Test (SUT). Dashed nodes indicate other equipment, i.e. support nodes, required to execute at least some of the tests. The latter nodes are considered not to be part of the SUT.

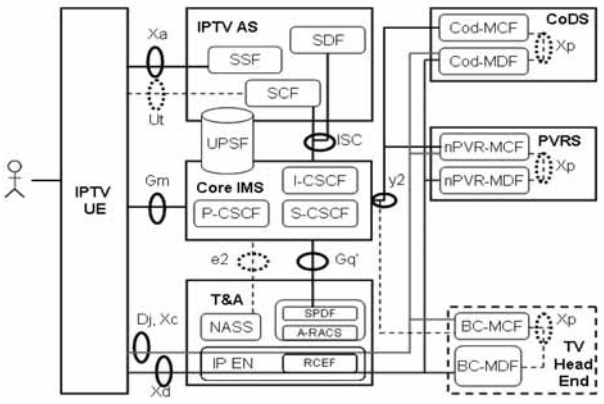

Fig. 2 IMS-based IPTV test architecture

Interoperability Test Description		
Identifier:	TD_IMS_IPTV_IM_0001	
Summary:	User sends an instant message through OMA Instant Messaging	
References:	ETSI TS 182 027 [1] clause 9.3.1; ETSI TS 183 063 [2] clauses 5.1.17.1 & 5.3.16.1	
Configuration:	CF_IMS_IPTV	
Required Equipment:	IPTV aware UE, Core IMS, IPTV AS	
Pre-test conditions:	• UE supports OMA Instant Messaging • UE is registered in Core IMS using userIPTV_priv identity	
Test Sequence:	**Step**	
	1	User registers to OMA Instant Messaging service
	2	User sends "Available soon?" IM
	3	Verify that UE receives SIP 200 OK
Conformance Criteria:	**Check**	
	1	Message exchange follows the below table

Step	Direction					Protocol	Comment
	User	UE	T&A	CORE	AS		
1	→						User registers to OMA Instant Messaging service
2		→					UE sends SIP REGISTER request including OMA feature tag [14] to CORE via Gm
3				→			CORE sends SIP REGISTER request to AS via ISC
4				←			AS sends SIP 200 OK response to CORE via ISC
5		←					CORE sends SIP 200 OK response to UE via Gm
6	→						UE sends "Available soon?" IM
7		→				SIP	UE sends SIP MESSAGE with specified header to CORE via Gm
8				→		SIP	CORE sends SIP MESSAGE with specified header to AS via ISC
9				←		SIP	AS sends SIP 200 OK response to CORE via ISC
10		←				SIP	CORE sends SIP 200 OK response to UE via Gm
11	←						UE receives SIP 200 OK

Fig. 3 Interoperability test description for Instant messaging sending

In the ETSI TS 186 020, the IMS-based IPTV interoperability test descriptions (TD) for systems composed of equipment by different vendors are defined. Each TD includes a test sequence describing user interactions with IPTV equipment as well as messages exchanged between IPTV equipment at selected standardized reference points.

3. Test Description – Instant Messaging Example

Instant messaging, often shortened to simply "IM" or "IMing," is the exchange of text messages through a software application in real-time. In this clause, we provide test description for Instant messaging in the context of TISPAN IPTV. On this example, we show how the IMS-based IPTV procedures with other IMS services are to be tested. The IPTV UE shall support OMA Instant Messaging according to [13]. It should be noted that SIP messages as 100 TRYING are not included in sequence diagrams below.

Interoperability test description for Instant messaging sending is illustrated in Fig. 3.

Interoperability test description for Instant messaging receiving is illustrated in Fig. 4.

4. Conclusion

IMS is expected to respond to and solve many of the industry's biggest technological challenges, including the lack of interoperability among operators who offer the same services and the inability of operators to take advantage of converged networks. IMS is a key component of the Next-Generation Networks and IPTV has been identified as one of the first services to be merged with Voice/Data communications on IMS architecture.

Only a few basic use cases of IPTV have been defined in TISPAN Release 2. On the other hand, TISPAN Release 3 covers many new use cases and features for IPTV and the possible interactions with Voice/Data communications such as Social TV, Incoming Voice call management and notification on TV screen. As a consequence, the update of the IMS-Based IPTV Interoperability Test specifications is an important step towards the fully inter= operable IPTV solution.

In this paper, we described the ETSI TISPAN approaches to IPTV. For one of them, the IMS-based IPTV, the testing framework was described. We selected the IM use case for test descriptions including an end user test sequence as well as high level message flows at key standardized reference points. Further work is still required to cover and provide test descriptions for all Release 3 features.

Interoperability Test Description		
Identifier:	TD_IMS_IPTV_IM_0002	
Summary:	User receives an instant message through OMA Instant Messaging	
References:	ETSI TS 182 027 [1] clause 9.3.1; ETSI TS 183 063 [2] clauses 5.1.17.1 & 5.3.16.1	
Configuration:	CF_IMS_IPTV	
Required Equipment:	IPTV aware UE, Core IMS, IPTV AS	
Pre-test conditions:	• UE supports OMA Instant Messaging • UE is registered in Core IMS using userIPTV_priv identity • UE is registered to OMA IM service (see TD_IMS_IPTV_IM_0001)	
Test Sequence:	**Step**	
	1	An Instant Message is required to be sent to UE
	2	Verify that UE displays the received IM
Conformance Criteria:	**Check**	
	1	Message exchange follows the below table

Step	**Direction**						**Protocol**	**Comment**
	User	**UE**	**T&A**	**CORE**	**AS**			
1					←			An Instant Message is required to be sent to UE
2				←				AS sends SIP MESSAGE to CORE via ISC
3			←					CORE sends SIP MESSAGE to UE via Gm
4			→					UE sends 200 OK to CORE via Gm
5				→				CORE sends 200 OK to AS via ISC
6		←						Verify that UE displays the received IM

Fig. 4 Interoperability test description for Instant messaging receiving

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A SURVEY ON CONTEMPORARY TERRESTRIAL AND HANDHELD MULTIMEDIA BROADCASTING NETWORKS

Digital video broadcasting (DVB) transmission technologies are suitable for wireless delivery of multimedia content both to static and mobile end-users. On the one hand, DVB-T standard has been adopted for digital terrestrial television (DTT) and it has been further developed into a newer standard, DVB-T2, which was recently finalized. On the other hand, DVB-H is a specification for the transmission of digital TV to handheld user terminals such as smart phones delivering IP-encapsulated data. The combination of satellite and terrestrial DVB technologies is DVB-SH which delivers wide coverage to mobile terminals and promises upgraded services to mobile and vehicle users via a satellite link. The objective of our research article is to present a survey on terrestrial, mobile and hybrid DVB multimedia networks. Emphasis will be given on quality of radio and service related aspects. A focus on performance evaluation methodology based mainly on measurement campaign will be proposed with a detailed presentation of experimental equipment for drive-test field measurements. Guidelines of measurements and analysis processes will be described. A public DVB network in the area of Athens will be our case study.

Keywords: *DVB-T/T2, DVB-H/SH, Multimedia Broadcasting, Quality of Service, Performance Evaluation, Measurement Campaign.*

1. Introduction

Digital video broadcasting (DVB) systems [1] are suitable for multimedia content delivery both to fixed and mobile end-users. In Europe, DVB-T (Terrestrial) standard [2] has been adopted for terrestrial digital wireless transmission. DVB-T has been further developed into newer standards such as DVB-H (Handheld) [3], now in operation, and DVB-T2 [4], which was recently finalized. Hybrid DVB-SH standard [5] has been developed in order to promise upgraded services to mobile and vehicle users via satellite link. The Geneva 2006 (GE-06) Agreement [6] regulates frequency usage in the broadcast bands of Europe, the digital transition to DVB and the analogue switch-off processes.

Transition from analogue TV to DVB is a complex feat. Frequency channels for new DTT programs must be allocated, safeguards must be taken to protect existing services from interference, equipment at transmission stations/towers must be changed, new digital services must be launched and viewers must be informed of the changes. The broadcast network must be planned, swapped and optimized. The role of network coverage and DVB quality measurements is absolutely important.

A crucial point of research is the quality of service (QoS) characterization of operating systems based on orthogonal frequency division multiplexing (OFDM) technology in a mobile radio environment. The objectives of our work are:

1. A short survey on audio-visual quality evaluation algorithms of the DVB user experience.
2. Key performance indicators (KPI's) that are critical for measurement like received RF signal power (RxLev), Carrier to Noise Ratio (CNR), Bit Error Rate (BER), Modulation Error Rate (MER), Frame Error Rate, Multi-Protocol Encapsulation (MPE) FER.
3. Experimental equipment for conducting indoor and outdoor measurements campaign.
4. QoS analysis procedures are proposed for DVB systems characterization.

Finally, we conclude with the importance of a field campaign in practical performance evaluation and optimization of a DVB radio network and possibly in benchmarking of DTT operators. The overall measurement methodology can be performed in the case of DVB-T2 or DVB-SH trial or in-deployment networks.

The rest of our research paper is organized as follows. The next section covers the major DVB systems for fixed (DVB-T/T2) and mobile/handheld (DVB-H/SH) reception. Section 3 gives an overview of key performance indicators and an experimental measurement campaign set-up and Section 4 provides QoS measurement campaign methodology. Finally, the conclusions and future work are drawn in Section 5.

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2. DVB Systems

2.1 Terrestrial DVB Standards

2.1.1 DVB-T Standard

In DVB-T standard, OFDM has been adopted in order to split the digital data stream into a large number of slower digital streams, each of which digitally modulate a set of closely spaced adjacent carrier frequencies. There are two choices for the number of carriers known as 2K-mode and 8K-mode with 1.705 4kHz carriers and 6.817 1kHz carriers respectively. Also, there are three available modulation schemes: QPSK, 16QAM, 64QAM.

DVB-T delivers data in a series of discrete blocks at the symbol rate. DVB-T is a COFDM transmission technique which includes the use of a Guard Interval. It allows the receiver to cope with strong multipath situations as it happens in case of mobile and vehicular environments. DVB-T also supports single-frequency network (SFN) operation, where transmitters carrying the same data operate on the same frequency in a geographical domain. Signals from each transmitter in the SFN need to be accurately time-aligned, which is done by sync information in the stream and timing at each transmitter referenced to GPS. The length of the Guard Interval can be chosen. It is a tradeoff between data rate and SFN capability. The longer the guard interval the larger is the potential SFN area without creating inter-symbol interference (ISI).

Specification [7] gives details of source coding methods for MPEG-2 and, more recently, H.264/MPEG-4 AVC [8] as well as audio encoding systems. Di Bari R. et al. in [9] conducted a measurement campaign for mobile DVB-T/H system.

2.1.2 DVB-T2 Standard

DVB-T2 [10] is the second generation of DVB-T and uses also OFDM modulation with a large number of sub-carriers delivering a robust signal. DVB-T2 uses LDPC (low density parity check) coding combined with BCH (Bose-Chaudhuri-Hocquengham) coding. The key DVB-T2 points are:

1. *Rotated Constellations,* which provide significant additional robustness in difficult radio channels.
2. *Multiple Physical Layer Pipes* that allow separate adjustment of the robustness of each delivered service within a channel to meet the required reception conditions (e.g. in-door or rooftop antenna).
3. *Alamouti coding* which is a transmitter diversity method improves coverage in small-scale SFNs.
4. *Extended interleaving,* including bit, cell, time and frequency interleaving.
5. *Future Extension Frames* allow the standard to be compatibly enhanced in the future.

As a result, DVB-T2 supports a much higher data rate than DVB-T as well as a much more robust signal. Eizmendi I. et al. in [11] conducted HDTV field trials using DVB-T and DVB-T2 broadcasting systems.

2.2 Handheld DVB Standards

2.2.1 DVB-H Standard

DVB-H is a technical specification for the transmission of digital TV to handheld receivers such as mobile telephones and PDAs. It is designed to enable the efficient delivery of IP-encapsulated data over terrestrial networks. DVB-H is closely related to DVB-T and includes modifications dealing with data broadcasting, service information, mobile TV, video streaming in general and file downloads, all targeted at handheld receivers that would operate with a limited battery life and in difficult reception conditions. The most important of these were that there should be a significant power saving in the receiver compared to DVB-T, excellent performance and robustness in a cellular environment, and enhanced support for single antenna reception in single frequency networks (SFNs).

DVB-H [12] is an extension of DVB-T with some backwards compatibility. It uses a mechanism called multi-protocol encapsulation (MPE), making it possible to transport data network protocols on top of MPEG-2 transport streams. A forward error correction (FEC) scheme is used in conjunction with this to improve the robustness and thus mobility of the signal. In addition to the 2K and 8K modes available in DVB-T, a 4k mode is added to DVB-H giving increased flexibility for network design. Another essential element of DVB-H is Time Slicing, the main technique used to achieve the required power savings. Each individual TV service in a DVB-H signal is transmitted in bursts allowing the receiver to go into sleep mode, only waking up when the service to which it is "*tuned*" is transmitted. Statistical multiplexing is also possible in DVB-H, ensuring optimum use of bandwidth to deliver services. DVB-H is designed for use in Bands III, IV and V as well as L-band.

2.2.2 DVB-SH Standard

DVB-SH [13] is the name of a transmission system standard designed to deliver video, audio and data services to vehicles and handheld devices. The key feature of DVB-SH is that it is a hybrid satellite/terrestrial system that allows the use of a satellite to achieve coverage of large regions or even a whole country. In areas where direct reception of the satellite signal is not possible, terrestrial gap filler can be used seamlessly to provide coverage. It is designed to use frequencies below 3GHz, typically around 2.2GHz [14].

The combination of a satellite and a terrestrial complement delivers wide coverage to mobile terminals which implement the TDM and OFDM modes of SH, a combination of SH and DVB-H, or simply the OFDM mode of DVB-SH operating in SFN. Key to deployment will be DVB-SH's interface with the existing DVB-IPDC layer and the services based on it. Cohen M. et al. in [15]

tested through various trials, including terrestrial only reception, satellite only reception and hybrid reception.

3. Quality Measurements of DVB Radio Systems

3.1 Multimedia Quality of Experience

3.1.1 Audio Quality Evaluation

Perceptual Evaluation of Audio Quality (PEAQ) algorithm was standardized by ITU-R for objectively measuring perceived audio quality. PEAQ simulate perceptual properties of the human ear and integrate multiple model output variables (MOV) into a single metric. PEAQ results principally model mean opinion scores (MOS).

Perceptual evaluation of speech quality (PESQ) algorithm supports only narrowband (4kHz bandwidth) speech signals while it does not support wideband applications used in contemporary multimedia communication systems.

Perceptual objective listening quality analysis (POLQA) is the new standardized algorithm for fixed, mobile and IP based networks. POLQA has been selected to form the new ITU-T voice quality testing standard, P.863, and will be used with HD Voice, 3G and 4G/LTE.

3.1.2 Visual Quality Evaluation

Objective video evaluation techniques are mathematical models that approximate results of subjective quality assessment, but are based on criteria and metrics that can be measured objectively and automatically evaluated by a computer program. Objective methods are classified based on the availability of the original video signal, which is considered to be of high quality (generally not compressed). Therefore, they can be classified as a) Full Reference methods (FR) b) Reduced Reference Models, and c) No Reference Models.

Recently, ITU-T standardized perceptual visual quality measurement techniques for multimedia services over digital cable television networks in the presence of a reduced bandwidth reference, J.246 [16], as well as objective perceptual multimedia video quality measurement in the presence of a full reference, J.247 [17].

A no-reference model has been proposed by ITU. Video quality MOS_{visual} is computed using the video quality parameters:

$$MOS_{visual} = 1 + I_{coding} \times \exp\left(-\frac{P_{plv}}{D_{Pplv}}\right) \quad (1)$$

where
- I_{coding} represents the basic video quality affected by the coding distortion under a combination of video bit rate (Br_V [kbit/s]) and video frame rate (Fr_V [fps]).
- P_{plv} is the packet loss rate,
- $Dplv$ is the packet loss robustness factor due to packet loss.

3.1.3 Multimedia Quality Evaluation

A subjective overall audiovisual quality, MOS_{av}, assessment method for multimedia applications is a mapping across laboratories from the separate audio, MOS_a, and video, MOS_v, quality was found to be [18]:

$$MOS_{av} = (c_1 + MOS_a) \times (c_2 + MOS_v) \quad (2)$$

Total multimedia quality index, MOS_{mm}, can be computed by the audio-visual delay, MOS_d, according to the following equations:

$$MOS_d = c_3 + c_4 \times \exp(c_5 \times d/c_6) \quad (3)$$

$$MOS_{mm} = (c_7 + MOS_{av}) \times (c_8 + MOS_d) \quad (4)$$

where $c_i \forall \in [1,8]$ are constants, and d is the "*lip-sync*" delay.

Recently, Singh KD and Rubino G. in [19] proposed no-reference QoE monitoring methods in DVB-H networks.

3.2 Radio Quality Aspects

Measurements guidelines for quality aspects in DVB-T systems are described in ETSI TR 101 290 [20]. Analytically:

3.2.1 System availability

The system availability describes the long-term quality of the complete digital transmission system from MPEG-2 encoder to the measurement point.

3.2.2 Link availability

The link availability describes the long term quality of a specified link in a digital transmission chain. It could be used as a quality of service parameter in contracts between network operators and program providers.

3.2.3 RF/IF signal power

Level measurement is needed to set up a network. The signal power, or wanted power, is defined as the mean power of the selected signal as would be measured with a thermal power sensor. Care should be taken to limit the measurement to the bandwidth of the wanted signal. When using a spectrum analyzer or a calibrated receiver, it should integrate the signal power within the nominal bandwidth of the signal.

3.2.4 Noise power

Noise is a significant impairment in a transmission network. The noise power (mean power), or unwanted power, is measured with a spectrum analyzer (out of service) or an estimate is obtained from the IQ diagram (in service). The noise level is specified using either the occupied bandwidth of the signal, which is equal to the symbol rate.

3.2.5 BER before RS decoder

The Bit Error Rate (BER) is the primary parameter which describes the quality of the digital transmission link.

3.2.6 Bit error count after RS

BER measure whether the MPEG-2 TS is quasi error free.

The same principle as used for the "Out of service measurement" of the "BER before the Reed-Solomon decoder", with the modification that the result is presented as an error count rather than a ratio. The receiver only has to compare the received TS packets with the Null packets.

3.2.7 I-Q signal analysis

Assuming:
- a constellation diagram of M symbol points; and
- a measurement sample of N data points, where N is sufficiently larger than M to deliver the wanted measurement accuracy; and
- the co-ordinates of each received data point j being $I_j + \delta I_j$, $Q_j + \delta Q_j$ where I and Q are the co-ordinates of the ideal symbol point and δI and δQ are the offsets forming the error vector of the data point.

Modulation Error Ratio (MER) and the related Error Vector Magnitude (EVM) are calculated from all N data points without special pre-calculation for the data belonging to the M symbol points.

With the aim of separating individual influences from the received data, for each point i of the M symbol points the mean distance d_i and the distribution σ_i can be calculated from those δI_j, δQ_j belonging to the point i.

From the M values $\{d_1, d_2, ..., d_M\}$ the influences /parameters:
- origin offset;
- amplitude Imbalance (AI); and
- quadrature Error (QE),

can be extracted and removed from the di values, allowing to calculate the Residual Target Error (RTE) with the same algorithm as the System Target Error (STE) from $\{d_1, d_2, ..., d_M\}$.

From the statistical distribution of the M clouds parameters:
- phase jitter; and
- CW interferer,

may be extracted. The remaining clouds (after elimination of the above two influences) are assumed to be due to Gaussian noise only and are the basis for calculation of the signal-to-noise ratio. The parameter may include – besides noise – also some other disturbing effects, like small non-coherent interferers or residual errors from the equalizer. From the SNR value the Carrier/Noise value can be estimated.
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$$SNR = 10 \times \log_{10} \left\{ \frac{\frac{1}{N}\sum_{j=1}^{n}(I_j^2 + Q_j^2)}{\frac{1}{N}\sum_{j=1}^{n}(\sigma I_j^2 + \sigma Q_j^2)} \right\} \quad (5)$$

Fig.1 depicts constellation diagrams for a double diversity receiver in case of excellent, good and bad coverage conditions.

- *Modulation Error Ratio (MER)*

MER provides a single "figure of merit" analysis of the received signal. This figure is computed to include the total signal degradation likely to be present at the input of a commercial receiver's decision circuits and so give an indication of the ability of that receiver to correctly decode the signal.

It should be reconsidered that MER is just one way of computing a "figure of merit" for a vector modulated signal. Another "figure of merit" calculation is Error Vector Magnitude (EVM). It is also shown that MER and EVM are closely related and that one can generally be computed from the other. MER is the preferred first choice for various reasons.

$$MER = 10 \times \log_{10} \left\{ \frac{\sum_{j=1}^{n}(I_j^2 + Q_j^2)}{\sum_{j=1}^{n}(\delta I_j^2 + \delta Q_j^2)} \right\} \quad (6)$$

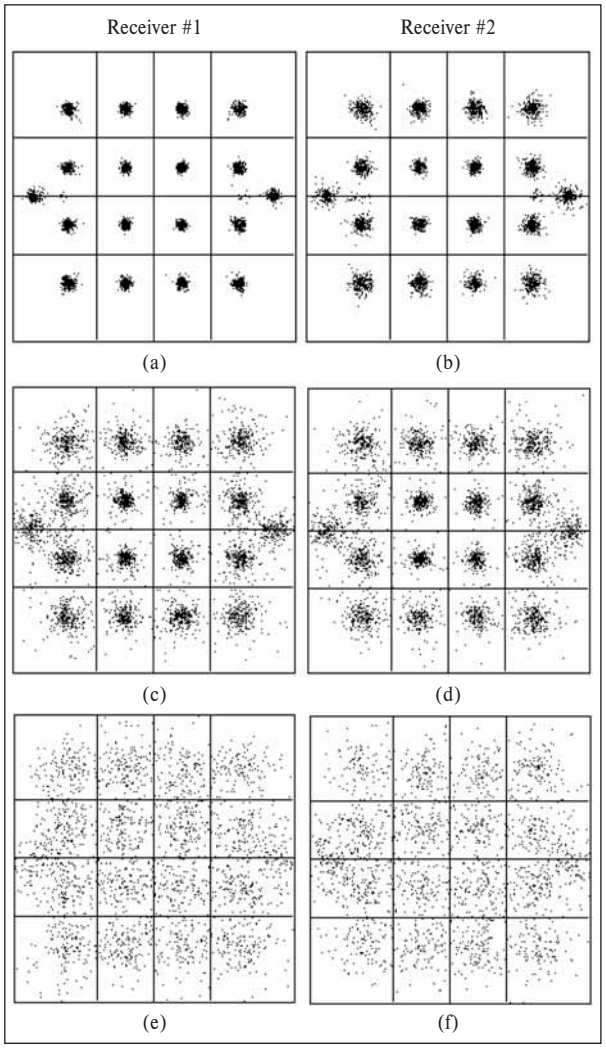

Fig. 1 Constellation diagram of the diversity DVB-T/H receiver with two RF input ports in case of excellent(a-b), good (c-d) and bad coverage (e-f)

- *System Target Error (STE)*

The displacement of the centers of the clouds in a constellation diagram from their ideal symbol point reduces the noise immunity of the system and indicates the presence of special kind of distortions like Carrier Suppression, Amplitude Imbalance, Quadrature Error (QE). STE gives a global indication about the overall distortion present on the raw data received by the system.

From the magnitude of the *M* Target Error Vectors calculate the mean value and the standard deviation (normalized to S_{rms}, defined as the RMS amplitude value of the points in the constellation), obtaining the System Target Error Mean (STEM) and the System Target Error Deviation (STED) as follows:

$$S_{rms} = \sqrt{\frac{\sum_{j=1}^{N}(I_j^2 + Q_j^2)}{N}} \tag{7}$$

$$STEM = \frac{\sum_{j=1}^{N}|\overline{d}_j|}{M \times S_{rms}} \tag{8}$$

$$STED = \sqrt{\frac{\sum_{j=1}^{n}|\overline{d}_j|^2}{M \times S_{rms}^2} - STEM^2} \tag{9}$$

- *Carrier suppression*

A residual carrier is an unwanted coherent CW signal added to the QAM signal. It may have been produced by DC offset voltages of the modulating I and/or Q signal or by crosstalk from the modulating carrier within the modulator.

- *Residual Target Error (RTE)*

The RTE is a subset of the distortions measured as System Target Error (STE) with influences of Carrier Suppression, Amplitude Imbalance, and Quadrature Error (QE) removed. The remaining distortions may result mainly from non-linear distortions.

- *Coherent interferer*

Coherent interferers are usually measured with a spectrum analyzer. From the statistical distribution of the clouds, the amplitude of the interferer can be calculated if it is above a certain limit.

4. Measurements System and Campaign

4.1 Measurements System

The functional block diagram of *R&S TSM-DVB* [21] (see Fig. 2) highlights two receiver channels in the form of any of the following:

- A diversity receiver with two different antennas connected to the two RF inputs that are 50Ω unbalanced.
- Two high-performance tuners that are compatible with VHF (Ch 5 - 12) and UHF (Ch 21 - 69). Channel bandwidth could be 5 MHz, 6 MHz, 7 MHz or 8 MHz. The RF input is 50 Ω unbalanced.
- Two discrete DVB-T demodulators suitable for mobile, portable and stationary applications.
- Two asynchronous serial interfaces (ASI).
- A configurable complex programmable logic device (CPLD) which is used for MPEG stream switching.

The measurements system was installed in a vehicle as it is shown in Fig. 3. The two antennas were racked on the roof and the RF cables were carefully driven through a window. A netbook

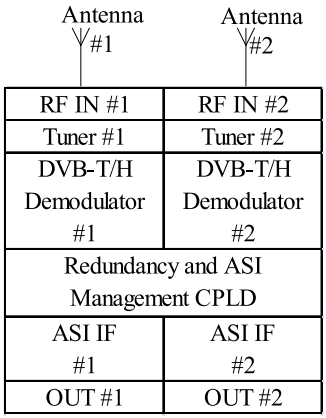

Fig. 2 Functional block diagram of the DVB-T/H diversity test system for drive-test measurement campaign

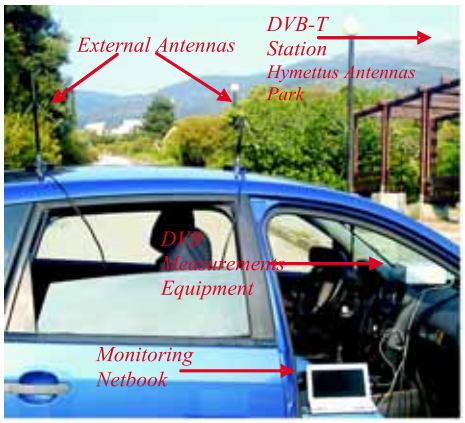

Fig. 3 Details of the measurements equipment installation before drive-test campaign

was used for measurements system control and measurements data storage. The power supply of the netbook and the drive-test equipment was directly from the vehicle's battery. All measurements were triggered with an external GPS receiver.

4.2 Measurements Campaign

- *Measurements Data Handling.* A set of processes shall be defined for measurements data handling. During a measurement campaign, large sets of data are going to be produced in a daily basis. The measurement team shall organize the produced measurements and deliver them to the analysis team to take into account security and integrity issues. No measurement data shall be lost or damaged. Data handling must be reliable end-to-end. Back-up process shall be defined by the analysis team.
- *Verification Tests.* A set of verification tests shall be executed before measurement campaign. A crucial point is the performance of antennas as well as the feeding RF cables. These components have to be in-lab tested via a network analyzer in order to characterize indicators like return or reflection loss (*RL*) of signal power from the reflection caused at a discontinuity in the RF cable, the reflection coefficient (Γ), or alternatively the standing wave ratio (*SWR*) by using a SWR meter.
- *Route Planning.* A route plan shall be defined before a measurement campaign starts. A measurement campaign is usually a long-time drive-test where urban, suburban and rural routes as well as major motorways, islands and touristic areas are to be measured. It is impossible to measure all around the territory. So, a representative subset of routes shall be chosen under population and geographic criteria. A time schedule shall be agreed according to the route plan. Besides, hotspots like shopping centers, airports, ports, railway stations and central town squares should be included to the schedule for static measurements.
- *Technical reports delivery.* Technical reporting is a crucial process during a measurement campaign of DVB radio networks. Analysis may be performed by type of route (town, motorway, touristic area, or island) as well as by wide geographic area (prefecture, region, allotment). A final executive/summary report shall be delivered at the end of the campaign. It is important to select representative and comparable key performance indicators (KPIs) for technical reporting. Post processing analysis of the KPIs is usually based on statistics. Results presentation is usually performed using coverage maps, probability and cumulative density function (PDF/ CDF) diagrams and pivot tables as well.

5. Conclusions

The scope of our paper was to propose a methodology for QoS characterization of real world digital video broadcasting (DVB) networks taking into account audio-visual quality. DVB standards for fixed terrestrial and mobile handheld transmission have been already commercially adopted for multimedia broadcasting deployment. Offspring DVB-T2 and DVB-SH technologies will be adopted by operators for broadband multimedia services. Analytically, we presented critical QoS indicators both for radio coverage and performance assessment. An experiment setup for DVB measurement campaign was described based on special laboratory equipment with diversity reception capability. Measurements analysis procedures were proposed for performance evaluation and of radio broadcasting networks.

Acknowledgements

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IMPLEMENTATION OF IMS TESTBEDS USING OPEN SOURCE PLATFORMS

The IP Multimedia Subsystem (IMS) is a control architecture framework for the Next Generation Network (NGN). It is expected that this new network architecture will provide a new kind of IP multimedia services delivered in network converged environments. The NGN and IMS, even after the years of a standardization effort, are still hot networking topics. One of the communities, which adopt emerging communications standards into their products fast, is the open source community. In present days, there are a few open source products, which allow building the IMS environment. However, there are at least two domains for which they are especially helpful. It is teaching/learning of the IMS architecture and testing service deployment in a nearly real environment. This article provides a survey of available open source products. It also describes the experience with the implementation, establishment and running an IMS testbed, which has been built up on an open source client and server entities. At the server side, the focus is on the Kamailio IMS platform.

Keywords: *IMS, open source, Kamailio, testbed*

1. Introduction

Next Generation Network (NGN) presents an interesting concept which merge different communications paradigms, wireline and wireless, mobile and Internet. The idea behind the NGN is to operate a universal communications platform covering existing network heterogeneity; which is multiservice, multiprotocol, multiaccess, IP-based and providing feature rich multimedia services. The fixed-mobile converged network enables to create a single communication environment allowing its users a unified access to their services from any fixed or mobile terminal via any access network.

NGN framework was introduced by the International Telecommunication Union–Telecommunication Standardization Sector (ITU-T), and ETSI (European Telecommunications Standards Institute). The key component of the NGN is IP Multimedia Subsystem (IMS) control layer, formerly defined by the 3rd Generation Partnership Project (3GPP) for mobile networks of 3rd generation [1, 2] and incorporated then into the NGN. IMS allows the move from circuit-switched services toward IP packet-based services and allows building up "All-IP" communications architecture with all of its benefits [3]. IMS allows providing of multimedia services in secure, chargeable environment with different levels of quality guarantees. IMS specifies a set of standardized entities and interfaces and allows building different kinds of integrated innovative services, which should combine features of legacy telecommunication and IP services. This concept brings a new Internet and IT way of the service creation approach to the development of innovative and attractive services.

Prototyping and delivering of such multimedia services or trial concepts is a challenging task which requires testing environments that allow understanding and evaluating of the behavior and impacts of proposed solutions. This should be one of many factors which help to speed up IMS adoptions. One of the communities, which adopt emerging communications standards into their products very fast, is the open source community. There are a few open source products which allow building up IMS testing/teaching environment to support learning models, service and components development. This article provides the survey of existing open source products which are suitable for IMS testbed establishment. It also describes the experience with implementation, establishment and running an IMS testbed which has been built up on open source client and server entities.

The paper is organized into five chapters. Chapter two provides the overview of NGN and IMS system architecture, chapter three analyzes key open source solutions suitable for establishing an IMS testbed, chapter four provides a description of our approach and chapter five concludes the article.

2. Next Generation Networks and IP Multimedia Subsystem

2.1. NGN architecture overview

The objective of NGN is to enable convergence of different fixed and mobile networks whose central component is the IMS.

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The NGN functional reference architecture [4] is designed as a system of horizontally oriented layers consisting of a number of cooperating subsystems. The subsystem design enables future subsystems to be added extending the main functionality and services, (for example, an IPTV in NGN Release 3). Each subsystem is specified as a set of entities interconnected through related interfaces. The essential NGN specification defines the NGN transport layer and the NGN service layer. The NGN transport layer provides IP connectivity for NGN users (UE) and hides the transport technology of the access and core networks below the IP. The transport layer is divided into two sub-layers, the transport processing functions sub-layer and the transport control sub-layer. In general, transport processing functions provide packet forwarding and routing functions together with more specific functions such as media processing/transcoding, relaying, IP level interconnection, resource control, etc. The transport control sub-layer provides attachment control (addressing, access authentication, authorization, location management) provided by the Network Attachment Subsystem (NASS) and functions responsible for the implementation of procedures and mechanisms handling policy-based resource reservation (Quality of Service (QoS), Network Address Translation (NAT)) and admission control provided by the Resource and Admission Control Subsystem (RACS) [4]. NGN service layer comprises a number of sub-systems such as Core IP Multimedia Subsystem (IMS), PSTN/ISDN emulation subsystem (PES), IPTV subsystem, other subsystems and common components whose functionalities are shared by the related subsystems. Finally, there is a user equipment (UE) subsystem defined as: "One or more devices allowing user access to network services delivered by TISPAN NGN networks" [5]. As an example of the UE, a telephone set can be mentioned which has to be NGN-compatible to make the full use of NGN services. The UE is sending/receiving user data over the transport layer. At the signalling layer, the UE communicates with NASS for network authentication/configuration and with IMS for handling services.

2.2. IMS architecture overview

The IMS is a core component around which the NGN is defined (Fig. 1). The IMS is end-to-end IP control framework that allows applications and services to be supported seamlessly across all types of networks. The IMS architecture capabilities are specified in a set of 3GPP documents that define components, services and interfaces [1, 2]. ETSI TISPAN NGN [6, 7] adopted the 3GPP IMS [1] for the NGN network architecture [4]. From an architectural point of view, the NGN IMS subsystem, also known as "Core IMS", is a subset of the 3GPP IMS, where some entities such as Application servers (AS) and transport/media related entities (Multimedia Resource Function Processor function (MRFP), IP Multimedia Gateway Functions (IM-MGW)) are considered to be outside of the "core IMS" [7] (Fig. 2). From a functional point of view, the 3GPP IMS has been extended to support additional access network types, such as xDSL and WLAN. The control functionality of the Core IMS handles session signalling and comprises a number of distinct functions to process signalling flows, such as Call Session Control Functions (CSCF), Breakout Control Gateway

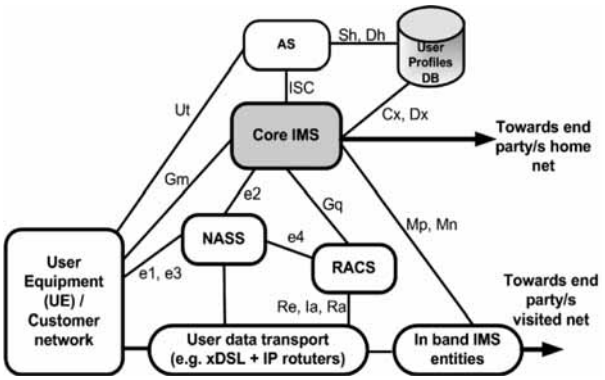

Fig. 1 IMS interfaces to the ETSI TISPAN NGN architecture

Functions (BGCF), Media Gateway Control Function (MGCF), Multimedia Resource Function Controller (MRFC) and Interconnection Border Control Function (IBCF). Using protocols such as SIP [8], Diameter and H.248 MEGACO, the different elements are able to establish requested subscriber services. The ETSI TISPAN is harmonizing with the 3GPP on any modifications or improvements that may be needed for the NGN/IMS.

2.3. Core IMS components

The functionalities of a core IMS are included in the Call Session Control Function (CSCF). A CSCF is a SIP server which processes the IMS signalling traffic in order to control IMS sessions. There are three types of CSCF, Proxy CSCF (P-CSCF), Serving CSCF (S-CSCF) and Interrogating CSCF (I-CSCF) (Fig. 2).

The P-CSCF is the entry point for signalling traffic flowing into the IMS. A user is allocated to a P-CSCF as a part of the registra-

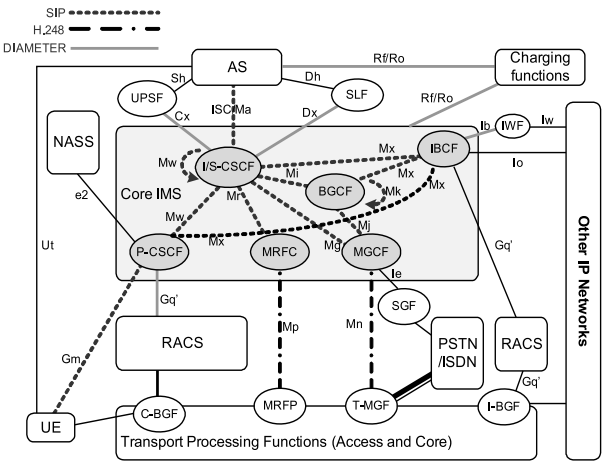

Fig. 2 Core IMS in the ETSI TISPAN NGN architecture

tion process. P-CSCF stays for a registered client on signalling traffic path for the duration of the session. P-CSCF should provide other functionalities as signalling compression, security association and resources management. The I-CSCF is usually located at the boundary of a domain that serves as a gateway to other domains, such as other service provider networks. I-CSCF may encrypt sensitive domain information (Topology Hiding Internetwork Gateway (THIG)) before forwarding the traffic. The S-CSCF is always located at a home provider network, provides the service logic which controls the delivery of requested service. The S-CSCF interacts with a profile database (HSS in a 3GPP environments, UPSF in a NGN environment) in order to determine user service eligibility by downloading the user profile; the S-CSCF is allocated for the duration of the registration.

3. Open Source IMS Testbed

Motivations behind the establishment of an IMS testbed are different; to expose complex communications concepts and components, to enable advanced NGN/IMS services development, to enable trial and performance testing, etc. For university education, there are all mentioned with an accent to popularize IMS technology on an advanced learning platform, to enable technology, protocol and services analyses and to provide safe and controlled environments to facilitate research and experimentations.

Similarly to SIP and IMS, an open source usually speeds up the implementation of emerging industry standards. Today's open source products allow the building of IMS testbed using standard hardware and operating systems (OSs) that provide cost effective solutions. This chapter provides a survey and feature analysis of suitable open source products available at the time of writing this paper.

3.1 Survey of IMS Core components

As defined in [1, 2, 6 and 7], the IMS consists of numerous entities mentioned before, including media and gateway controllers. It should provide services in secure, chargeable environments with different levels of QoS. This is a really challenging task which currently cannot be fulfilled by any of open source solutions. However, for establishing a simple IMS system with basic functionalities allowing call establishment and registrations, four components are necessary only, P-CSCF, S-CSCF, I-CSCF and a database (Home Subscriber Server – HSS) used to store user profiles. Therefore, main research and development effort is focusing on the control and application layer of the IMS, while the transport layer with QoS reservation concept, RACS and NASS subsystems, is currently omitted. IP connectivity is required as a fundamental prerequisite. Control layer components which perform signalling routing, authorization, registration and service provisioning are central parts of a testbed. To build up an IMS testbed, there are two open source solutions available, the Open IMS Core developed by the Fraunhofer Institute FOKUS [9] and Kamailio IMS extensions developed by the NG Voice [10].

3.1.1 Open IMS Core

Main motivation behind the development of the Open IMS Core, led by the Fraunhofer Institute FOKUS, was to provide an IMS core reference implementation suitable for IMS technology testing, service prototyping, technology research and many others. The Open IMS Core is neither designed nor suitable for commercial IMS architecture deployment. The Open IMS Core is an open source implementation (GPLv2 license) of IMS CSCFs signalling entities and the HSS [9] (Fig. 3) compliant with 3GPP specifications. The Open IMS Core is based upon open source solutions such as the SIP Express Router (SER) implementation of SIP Proxy server and the MySQL database. SER is characterized by powerful modular design and may act as a SIP registrar, a SIP redirect and proxy server. The Open IMS Core extends main IETF SIP functionalities of the SER express router with IMS modules (namely CDiameterPeer, IMS Service Control, Proxy-CSC, Interrogating-CSCF, Serving-CSCF modules [11, 12]) to provide core IMS signalling functionalities of IMS CSCF entities (called Open IMS CSCFs). As the IMS requires some information lookup (user and service profiles), a simple, lightweight version of an HSS entity (called Fokus HSS – FHoSS) has been developed and added to the

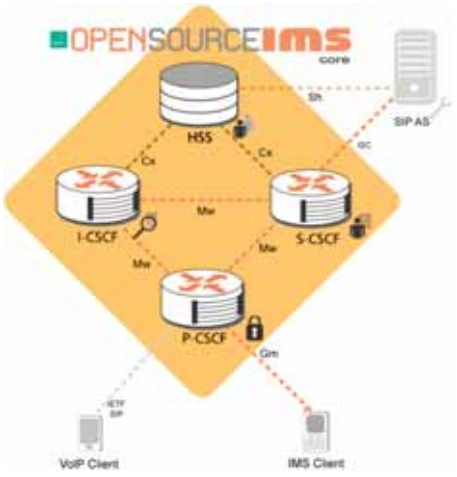

Fig. 3 The Fraunhofer Institute FOKUS Open IMS Core architecture

Open IMS Core project. The FHoSS is written in Java and uses the MySQL database. The FHoSS contains its own Java based Diameter stack which supports Cx, Sh and Zh IMS Diameter interfaces. The FHoSS provides functions such as storage of user profiles and service subscriptions, charging information, generation of authentication/authorization data, and a support of subscriber locations. The Open IMS Core provides an ISC (IMS Service Control) interface to allow the development of IMS applications or modifications of routing rules or events.

The Open IMS Core project provides quite comprehensive web documentation which describes the required steps for the testbed installation with basic IMS functionalities. The Open IMS

Core entities run over standard Linux based OSs. The installation requires compilation of installation packages, but the source codes contain the pre-prepared configuration files of CSCF entities. Main configuration requires small changes only related to the DNS names used in a testbed environment and port numbers on which CSCFs entities are listening. The user and service management is provided over the web-based FHoSS management graphical user interface (GUI).

The Open IMS Core project started on November 16, 2006. One of the goals of the project was to make the Open IMS Core an enabler for other testbeds and to create a community of IMS testers and developers that are using the Open IMS Core for their projects. Having analyzed the information available through the Internet, we found that this goal was met successfully [11, 13]. However, the last modification of the project was during 2009 and since that time the project has not been actively maintained.

3.1.2 Kamailio based IMS solution

As mentioned before, the Open IMS Core project is a perfect tool for the IMS playground establishment. However, there are several issues that could limit its future deployment. The Open IMS Core project is no more evolving neither supported now. Core CSCF entities of the Open IMS Core framework are based on an old version of the SER server (version 2.0 Beta).

The SER (SIP Express Router) server, nowadays known as the Sip-Router (since 2008), is a famous SIP proxy with a number of worldwide implementations and wide open-source community of developers, maintainers and supporters. The SER has been developed since 2001, and released in 2004 under the open source GPL license. The history behind the SER development is complicated, and leads to establishing several viable forked solutions, such as OpenSER in 2005 (renamed to Kamailio in 2008) and OpenSIPS in 2008. Nowadays, the developers of Kamailio and SIP-router projects are normalizing the situations and the merging of these two projects has been initiated. At present, Kamailio and SER should be considered as the same application (since release version 3.0). The OpenSIPS fork is still going its own development way.

Kamailio (former OpenSER) [14] is a very popular and viable SER based SIP proxy. The Kamailio server, released under the GPL license, is able to handle thousands of call setups per second. The Kamailio features include a support for asynchronous TCP, UDP and SCTP, secure communication via TLS for VoIP (voice, video), SIMPLE instant messaging and presence, ENUM, least cost routing, load balancing, routing fail-over, accounting, authentication and authorization against MySQL, Postgres, Oracle, Radius, LDAP, XMLRPC control interface, SNMP monitoring [14]. During 2011 the Kamailio IMS module extensions were introduced, which allow to build an IMS testbed with the latest Kamailio SIP server releases. The main developer of the Kamailio IMS extensions is Mr. Carsten Bock, a member of the Kamailio development team [10]. The Kamailio IMS module project aims to create an open-source based stable IMS infrastructure based on different GPL solutions [10]. As a basement for Kamailio IMS modules, former Open IMS Core modules were taken and ported for re-using with Kamailio. However, there are several issues identified. Since the Open IMS Core deployed all the IMS functionalities into a few modules, whose functionalities and programming code are partially duplicated by the Kamailio server, the development is progressing fast with many improvements (architecture re-designs, performance improvements, feature extensions, etc.). The future modules rebuild is being developed with the objective to define modular IMS modules design without duplicating functionalities of the Kamailio server and IMS modules, re-using functionalities of Kamailio modules wherever possible (for example, a dialog module), enhancing an IMS module wherever necessary, reusing Kamailio presence functionalities, and finally adding new features [15]. Actually, some improvements have been already provided, for example, support for Reg-Event, enhancements of a dialog module and RTP activity detection. Developers are currently working on other improvements (especially, on Diameter protocol). Kamailio IMS intended to provide high-performance, open-source and standards-compliant implementation of HSS for the use in IMS context. Due to lack of human resources the project was frozen. Instead, the Open IMS Core FHoSS is used. At present days, the Kamailio IMS project, although short-lived, allows to build up an operational IMS testbed. Kamailio based CSCF entities provide many features, many of them are fully open source based, standards compliant and providing stability and performance with full IPv6 support. The whole feature list is available at [10].

The Kamailio project provides wide, high-quality and up-to-date web based documentation flesh out with a lot of external internet documentation sources. The information about a Kamailio IMS testbed installation is documented in [10]. The knowledge base is also provided on the knowledge portal of the Department of information networks [16]. Kamailio IMS runs over standard Linux based OSs. Installation packages for 32bit debian operating systems are available through the package repository, while the other operating systems require their compilation. Configuration of Kamailio based IMS requires modifications of configuration files with correct DNS names of a testbed environment, modifications of port numbers on which CSCFs entities are listening and finally, modifications of Diameter configuration files with correct IP addresses and port numbers required for Diameter interconnection [16]. Comparing to the Open IMS Core installation process, Kamailio IMS configuration process is more difficult, which is caused by missing lines inside the Diameter configuration files which are required for establishing correct Diameter interconnection interfaces.

3.2 Survey of available IMS clients

IMS clients or end user equipment are entities through which users interact with IMS entities and perform signalling procedures and service invocation. IMS clients may range from clients that support registration and a voice call only up to clients that support voice/video calls, messaging and presence, IPTV, and etc. The UE connected to the IMS testbed should support AKAv1/2-MD5 authentication, IMS SIP signalling (PRACK, precondition), and

voice calls. Advanced IMS clients should support voice and video calls, messaging and presence, address books, advanced security functionalities and they should be NAT friendly. Up to now, there are several open source projects that have been working on IMS client development. Some of them have been already closed, some of them are still active. Table 1. provides a feature analysis of six currently identified IMS clients that are available under some of open source or free licenses and that are suitable for IMS testbed usage for different scenarios. Those are Boghe for Windows, iDoubs for iOS and Mac X OSs, IMSDroid for Android, myMonster and IMS Communicator, all of them are operating system independent as they are developed in Java. The UCT IMS client is intended for Linux users only.

Boghe is an IMS client for windows OSs which supports Rich Communication Suite (RCS). RCS is managed by the GSM Association (GSMA) with a goal to provide a set of fully interoperable rich services to be used inside the fixed-mobile environment and to speed up the evolution of mobile phone communication towards service rich communication. To become a RCS compliant, a client must at least support Enhanced Address Book (defined by the OMA), Enhanced Messaging (OMA), Content Sharing (GSMA) and File Transfer (OMA) services. A Boghe IMS client is based on a Doubango RCS framework which is shared with co-projects developing IMS clients for iOS and Android. A client provides user friendly GUI supporting comfortable configuration, many types of audio and video codecs. Comparing to other IMS clients, Boghe

IMS client feature analysis. Table 1.

IMS clients	Boghe IMS/RCS client	iDoubs	IMSDroid	myMonster TCS	UCT IMS client	IMS Communicator
www	code.google.com/p/boghe	code.google.com/p/idoubs	code.google.com/p/imsdroid	www.monster-the-client.org	uctimsclient.berlios.de	imscommunicator.berlios.de/
License	GPLv3	GPLv3	GPLv3	free, own	GPLv2	LGPL
Last release	2.0.91.683	2.0.184	2.0.481	0.9.19 TCS	1.0.13	0.70.605
Date	24.2.2012	20.9.2011	28.2.2012	2009	13.2.2009	8.7.2007
Operating system	Win	iOS,Mac OS X	Android	Multiplatform (Java)	Linux	Multiplatform (Java)
Multiple accounts	No	No	No	Yes	No	No
Registration	AKAv1/v2-MD5, MD5, basic	AKAv1/v2-MD5, MD5, basic	AKAv1/v2-MD5, MD5, basic	AKAv1/v2-MD5, MD5	AKAv1/v2-MD5, MD5	AKAv1-MD5, MD5
IMS Signalling	Yes	Yes	Yes	Yes	Yes	Yes
SigComp	Yes	Yes	Yes	No	No	No
Audio	Yes PCMA, PCMU, GSM, AMR-NB-OA, AMR-NB-BE, iLBC, Speex-NB	Yes G.722, G.729AB, AMR-NB, iLBC, GSM, PCMA, PCMU, Speex-NB, Speex-WB, Speex-UWB	Yes G729AB1, AMR-NB, iLBC, GSM, PCMA, PCMU, Speex-NB	Yes PCMA, PCMU, MCA	Yes GSM, PCMA, PCMU, MP2	Yes JMF codecs
Video	Yes MP4V-ES, Theora, H264, H263/+/++	Yes VP8, H264, MP4V-ES, Theora, H263/+/++	Yes VP8, H264, MP4V-ES, Theora, H.263, H.263-1998, H.261	Yes H.263, MP2T, H.263-,MPV, MP4V-ES	Yes -	Yes JMF codecs
Messaging/Chat	Page / Session mode	Page / Session mode	Page / Session mode	Page / Session mode	Page mode	-
Presence	Yes OMA, IETF	Yes OMA, IETF	Yes OMA, IETF	Yes OMA, IETF	Yes -	Yes -
XDMS/XCAP	Yes	Yes	Yes	Yes	Yes	No
Security	TLS, IPSec	-	TLS, IPsec	-	No	IPSec
IPv6	Yes	Yes	Yes	Yes	Yes	Yes
NAT Traversal	Yes STUN/TURN	Yes STUN/TURN	Yes STUN/TURN	No	No	Yes STUN
P-CSCF discovery	Fixed, DNS NAPTR+SRV	Fixed, DNS NAPTR+SRV	Fixed, DNS NAPTR+SRV	Fixed, SRV DNS	Fixed	Fixed
Other features	Enhanced Address Book, Content Sharing, File Transfer, Image share	-	File Transfer, Image Sharing, Video Sharing	File transfer, Group list management	IPTv (RTSP)	-

is the most recent initiative (released in 2011). The project is alive, the latest released version of the client is 2.0.85. Clients iDoubs (iOS, Mac X) and IMSDroid (Android) are very similar to a Boghe client, they are produced by the same developing team using the same Doubango developing framework [17] and they share most of the functionalities with a Boghe client.

The myMonster TCS IMS client is a feature rich client, based on the Telco Communicator Suite (TCS). TCS is a java based telecommunication framework from the myMONSTER toolkit (Multimedia Open InterNet Services and Telecommunication EnviRonment) developed at the Fraunhofer Institute FOKUS. The myMONSTER TCS supports 3GPP IMS, GSMA RCS and OMA service enabler access as Presence, XML Document Management (XDM), SIMPLE IM-based instant messaging and other functions. A client provides a user friendly GUI, it can be configured quite easily and supports multiple accounts. The latest released version is 0.9.19 from 2009 for Windows mobile, Windows and Linux OSs. Since that time, the project has not been actively evolving.

The UCT IMS client is developed by the University of Cape Town, South Africa. UCT IMS client is intended for Linux users only and it is intended especially for developers. It contains a message debug window and two preconfigured testing accounts for Alice and Bob. The latest version is 1.0.13 released in 2009. Again, since that time it has not been actively evolving. However, the support for an IPTv service with RTSP provides an interesting feature.

The IMS communicator is a Java based IMS client (JAIN), which does not require installation. Originally, it was developed by the PT Innovacao to support IMS service testing and IMS service development. The GUI interface is simple, however, client configuration is quite comprehensive and requires deeper knowledge. To make configuration more friendly, it provides a simplified configuration wizard. A client contains hard coded DNS resolving, and without java files modification a client is useless. The IMS Communicator has not been updated since 2007, and the project itself is not active anymore. Active tests executed over an IMS testbed show the ability to use vanilla IETF SIP clients for simple voice/video call scenarios.

For the reasons mentioned above, we recommend to use Boghe, followed by the myMonster TCS as the most feature rich and promising IMS clients.

4. Deploying an IMS testbed

To popularize IMS technology as an advanced learning platform, an enabler technology for a future protocol and services analyses, research and experimentations, we have analyzed the possibilities of using an open source platform for building up an IMS testbed. The IMS platform should provide a living environment that may be used for learning master degree courses offered by our department. It also should allow an access to the IMS technology and provide basic IMS services accessible from IMS clients. The testbed should be established with low costs based on open source software whenever possible. The IMS testbed should be open and extendible with new and innovative services and communications components.

Having used open source components mentioned above, we built our laboratory prototype environment which fulfills all these requirements. Even though in our research we are focusing on a Kamailio SIP server [14], the topology simulates two IMS providers and it is based on the both open source IMS projects. The first provider is based on the Kamailio IMS; the second one uses the Open IMS Core. The IMS testbed consists of core IMS entities only (CSCFs and FHoSSs). Our tesbed is realized with two servers (even though one of them is required only) and several software IMS clients connected over a LAN network. The first server is the departmental DNS server which keeps DNS NAPTR and SRV records required for the both virtual IMS providers. An example of the DNS configuration for virtual provider called IMS1 (DNS FQDN domain ims1.sip.uniza.sk) is described in Fig. 4.

```
$ORIGIN ims1.sip.uniza.sk.
@               NAPTR 10 100 "S" "SIP+D2U" "" _sip._udp.pcscf
@               NAPTR 20 100 "S" "SIP+D2T" "" _sip._tcp.pcscf
pcscf           IN   A          158.193.139.40
_sip.pcscf      IN   SRV  0  0   5060 pcscf
_sip._udp.pcscf IN   SRV  0  0   5060 pcscf
_sip._tcp.pcscf IN   SRV  0  0   5060 pcscf
icscf           IN   A          158.193.139.41
_sip            IN   SRV  0  0   5060 icscf
_sip._udp       IN   SRV  0  0   5060 icscf
_sip._tcp       IN   SRV  0  0   5060 icscf
scscf           IN   A          158.193.139.42
_sip.scscf      IN   SRV  0  0   5060 scscf
_sip._udp.scscf IN   SRV  0  0   5060 scscf
_sip._tcp.scscf IN   SRV  0  0   5060 scscf
hss             IN   A          158.193.139.43
presence        IN   A          158.193.139.37
```

Fig. 4 DNS records required for an IMS testbed

Thanks to Xen server virtualization technology, the second server is hosting all eight virtual servers with CSCFs and HSSs IMS control entities (Fig. 5). As the OS platform, 32 bit linux Debian OS release Squezze was used, and all testbed components are based on open source software (Kamailio IMS, Open Core IMS, Mysql, Apache, Boghe, Monster IMS clients, RTPproxy). To support a call establishment behind the Network Address Translator (NAT), the RTPProxy media proxy is used.

Over the testbed, we have provided several testing interconnection scenarios for different basic services such as audio/video calls, file transfers and messaging, different IMS clients, and simulating intra and inter domain connectivity. The signalling message exchange have been captured, analyzed and evaluated to proof the interconnectivity. Some scenarios demonstrated the applicability of "vanilla" IETF SIP clients which were able to register into an IMS domain and to make simple calls. The quality of IMS client implementations differs and the tests of interoperability show that

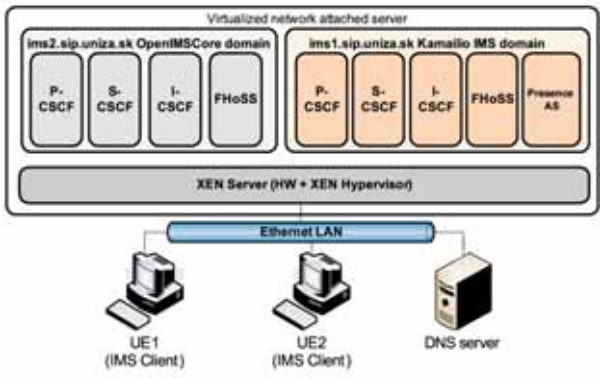

Fig. 5 IMS testbed scenario

while the basic connectivity is usually successful, the interoperability of clients using advanced services is not. Similar results are declared in [18]. In these cases a deeper structural testing and diagnostic is needed. We can recommend the Boghe IMS client following by Monster TCS as the most appropriate IMS clients. The testbed is possible to be extended and to be redundant with the deployment of more S-CSCF and P-CSCF servers. Both IMS solutions provide basic IMS feature set, with different UE authentication methods, as Digest-AKAv1/v2-MD5, Diget-MD5, SIP Digest. Additionally, both solutions support transport (Transport Layer security – TLS) and network layer (IP Security – IPSec) security mechanisms. As a result of our experiments, several articles have been published at the web site of the Department of InfoCom Networks [16].

4.1 Advanced services and other extendibility

The development of new IP services is one of the assets of IMS. We have shown that the both IMS solutions can be extended with new components providing new features and services, for which other open source solutions may be used. Our testbed was simply extended with a presence service based on the Kamailio SIP server that was used as the Presence AS. Kamailio provides an embedded XCAP server which supports the IETF SIMPLE messaging and presence standards. The Kamailio SIP server may be used as a standalone AS server as it supports Lua programming language for service creation. We identified other open source projects such as Mobicent Open Source Cloud Communications framework [19] (it consists of many subprojects as the Presence service for OMA SIMPLE, JAIN servers, SIP servlet server, media server and so on), the OpenVCS for video conferencing [20], the UCT Advanced IPTv for IPTV service [21], popular B2BUA as Asterisk and FreeSWITCH as media servers.

Tests of SIP based presence and messaging documents different level of standards adoption. The SIMPLE page mode and presence status indication work successfully. More advanced services such as the session mode which uses MSRP protocol or contact lists (address book) services (XCAP with XDM) reached poor results.

5. Conclusions

The deployment of the both open source IMS solutions with basic functionalities is quite easy and fast. It can be deployed with relatively low costs (apart from the costs required for a hosting hardware platform). Documentation is publically available; this, of course, accelerates research possibilities of universities, operators and vendors and enables the development of new IMS applications and testing new communications components. We tested the deployment of IMS testbeds which will be used in our learning and scientific practice. We also compared the two IMS frameworks; the Open IMS Core provides a bigger range in terms of installed solutions and users. This may be a result of its exclusive position for years. The Open IMS Core has simpler installation procedure than the Kamailio IMS. It has more detailed information database spread around the world describing many interesting extendibility solutions. However, the Open IMS core has not been actively evolving. Moreover, technological improvements of an underlying Kamailio SIP Server component against the SER SIP server 2.0, which is used as the basement of CSCF IMS entities, are notable. The technological advance and improvements of Kamailio SIP (actually, version 3.2.2), on which the Kamailio IMS was built, provides better future prospects. The Kamailio SIP server is a viable open source solution with many developers, testers and many installations. The number of new Kamailio IMS based installation will grow in a near future and will exploit a rich feature set of integrated solutions, based on Kamailio IMS and Kamailio AS. Even though the vision of open source projects is always difficult to be provided, the vitality of the Kamailio project is very promising.

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SPEECH CODER IDENTIFICATION USING CHAOTIC FEATURES BASED ON STEGANALYZER MODELS

In this study a steganalyzer model based on chaotic features is adapted to codec identification problem. As conventional steganalyzers detect suspected packages buried in bitstreams using statistical analysis, the proposed "speech coder identification" model also reveals the statistical differences of the outputs produced by different speech codecs. Essentially, the design of a coder/codec identifier is equivalent to a classifier training where the chaotic features extracted from the suspicious bitstream samples are used as inputs. During training, weight values which dissociate the codec types best are investigated. Finally, performances of the trained classifiers are evaluated by using test sets. According to the test results, for a closed group which is formed from nine different output types, polynomial SVM classifiers have identified more than 97% of the samples correctly.

Keywords: *Coder identification; codec recognition; suspicious bit stream*

1. Introduction

Codec or coder identification which is used interchangeably throughout this manuscript can be defined as identifying the type of codec from an unknown bit stream obtained from Internet, transmitted through air or stored in some media. Generally the type of codec, which will be used in voice, audio or video transmission, is nominated during the channel establishment stage. Anyone, who is able to monitor the transferred data and aware of the connection control protocol, can reveal the type of codec that is being used. In the case of storage media, voice data is usually stored in files which start with a header composed of attributes like the type of codec used. In some cases however there may be too many alternative connection control protocols that must be taken care of or monitoring of the bit stream might start after connection is established. For the storage media, file headers may be missing, corrupted or fake. For scenarios like these, it could still be possible to find the correct definition of the codec type being used by just analyzing the recorded bit stream.

The main motivation of this study is to employ steganalyzers, in order to identify speech coders, using bits obtained from an unknown source. During the training stage of a steganalyzer, firstly the statistical properties, which dissociate data hidden samples from regular data are investigated. Secondly for the "best" dissociated properties the threshold values, which separate the classes, are calculated. From the nature of the classifier design, decisions are either true positive (TP), false positive (FP), true negative (TN) or false negative (FN). For the steganalysis of suspicious speech, Kocal et al. [1] have proposed steganalyzers that decide by processing chaotic features. The outputs of different speech – voice coders would also have different statistical properties and as a result the design methodology of a speech steganalyzer can be adapted to speech codec identification problem.

2. Codec Identifier

As speech codec identification or speech codec type recognition is introduced to the literature with this paper, the term "codec identifier" needs to be defined. Codec identifier is a new generation bit analyzing method that is used to recognize the type of codec being used in a flowing or a stored bit stream. Note that codec identification can be performed for any bitstream which belongs to speech, audio or video. In this study the proof of concept is carried out on speech.

3. Design principles of the codec identifier

Codec identifier aims to recognize the type of codec used in a suspicious bit stream. Recognition can be handled using several different ways which all benefit from some sort of distinguishing information. However collecting, extracting and transforming all possible types of useful information may not be practical or feasible. In order to develop an implementable speech codec identifier model, some design criteria need to be specified along with the assumptions of the bit stream.

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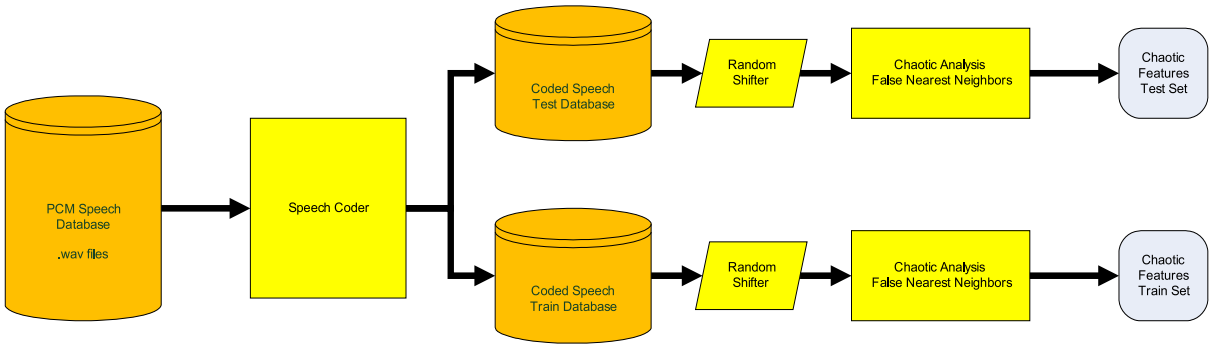

Fig. 1 Calculation of the chaotic features for a speech coder

Statistical Differences: It is assumed that different speech coders produce bit streams with different statistical properties.

Bit rate independency: Outputs of the coders are analyzed without any bit rate information. Different coders may produce different amounts of data in a unit amount of time, codec identifier proposed in this paper does not need this information, it only takes the output bit stream of the coder, divides bit stream into frames and analyzes these frames.

Bit allocation independency: Codec identifier analyzes output stream by only using the calculated chaotic features. The output of a speech coder is not a random sequence; bit fields that form the output bit stream should be grouped according to the information they convey, such as line spectral frequencies (LSF), pitch etc. The proposed method does not need or implicitly use this information. On the other hand, the order of grouping and/or logical relationships between these fields would have direct or indirect effects on the statistical properties and they should somehow reveal themselves in the calculated features.

4. Model Derivation from Steganalyzers

The idea behind the codec identification model design is quite simple: Any tool that is able to distinguish the statistical differences between the codec outputs should be able to detect the type of the codec used. Steganalyzers are sensitive to statistical differences and therefore codec identification should be possible by using existing steganalyzer models.

Codec identification is performed by using a general codec identifier model which is derived from the steganalyzer model given in [1]. In the steganalyzer model in question, statistical differences between two groups (group of samples which carry hidden data and group of samples which do not carry hidden data) are investigated in the chaotic feature domain. There are two types of features obtained from the evaluated samples: False Nearest Neighbors (FNN) [2] and Lyapunov exponents [3]. Existence of the hidden data may stretch any chaotic feature, since every data hiding method causes separation for these features and each data hiding method has its own specialized steganalyzer. The lengths NFNN FNN and NLY Lyapunov feature vectors vary according to dimension and delay [2, 3].

5. Feature Selection for Codec Identification

Codec identification is based on the statistical differences between outputs of the classified codecs. Chaotic features calculated from bit stream samples can be accepted as digest values of the bit stream samples. If the outputs can be identified according to the codecs in use, then these digest values should be separable according to the codec type being used.

In Fig. 1, calculation of test and train chaotic feature sets which belong to a speech coder type is described. First all of the samples stored in a speech database are coded by a speech coder and the output (coded samples) is divided into two smaller databases. Each sample in these databases is randomly shifted and pushed into chaotic analysis to produce its chaotic features. As all the samples are processed, two chaotic feature sets (test and train chaotic feature sets) are produced. Since the codec identifier is supposed to work on all possible portions of suspicious bitstream samples, where the beginning and end are unknown, for a fair testing scenario chaotic analysis should not be always calculated from the beginning of the coded sample. Random shifter simply throws away TR bytes from the beginning and shortens the sample.

In the case of codec identification, type of a codec used causes significant separations almost in every feature. Unlike steganalysis where specialized steganalyzers are trained, this phenomenon enables the design of a universal classifier which can potentially classify or identify all types of speech codecs and may give the correct classification by using only FNN as features.

As this work can be described as a proof of concept work, the relationship between the number of chaotic features (N_{FNN}), window sizes (The length of bitstream where FNN values are calculated for) and codec identification performance is not investigated in a detailed fashion. Therefore, all training and tests are performed for $N_{FNN} = 15$ case., On the other side theoretically,

- If N_{FNN} increases, then
  - Identification performance increases,
  - Complexity increases,
  - Larger train sets are required.
- If N_{FNN} decreases, then
  - Identification performance decreases,
  - Complexity decreases,
  - Smaller train sets are sufficient.
- If the window size increases, then
  - Identification performance increases,
  - Complexity increases.
- If the window size decreases, then
  - Identification performance decreases,
  - Complexity decreases.
- If N_{FNN} increases while the size of the training set is kept unchanged
  - Identification performance increases up to an optimum $N_{FNN\ opt}$
  - Identification performance decreases after exceeding the optimum $N_{FNN\ opt}$

Moreover for most practical scenarios, N_{FNN} is certainly an application specific value and applying trial – error procedures may probably be the best way to determine its value.

In Fig. 2, 2D points, which are obtained after principle component analysis (PCA) of 15 dimensional FNN feature vectors calculated from 1024 byte samples (N_{FNN} = 15, N_{LY} = 0), are illustrated. For simplicity, only the outputs of four different types of speech codecs (AMR 4.5K [4], G.726 16K [5], G.726 24K and G.729 32K [6]) are taken into consideration. For each 1024 length speech sample, a FNN feature 15 item vector is calculated. Each item in the vector gives false nearest neighbor fraction for specific dimension and delay. The item's value is always positive and changes between 0 and 1. Then all the calculated FNN vectors are collected together in a collection matrix. The collection matrix is composed of 2048 vectors, which means it carries 512 vectors per codec. After that, 2048×15 collection matrix is projected onto (or converted to) a 2048×2 matrix. The values of the projected matrix alter between the values −1 and 1. Finally all the vectors (or now they are coordinates) belonging to each codec are shown

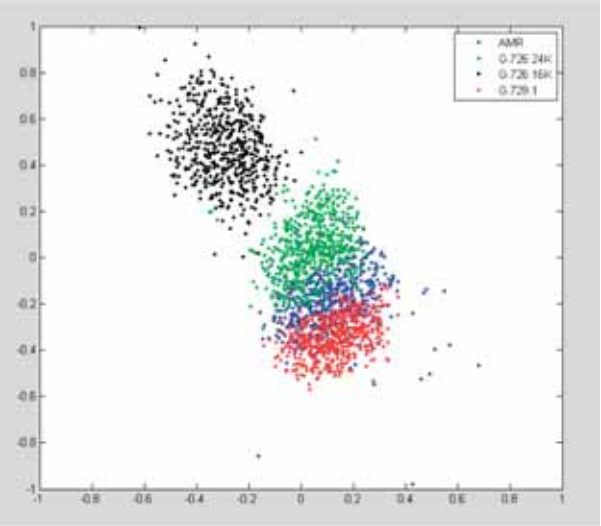

Fig. 2 Separation of projected feature vectors obtained from suspicious bit stream samples

with a different color. As illustrated in the figure, the chaotic features of four codecs are still separable from each other even after a 2-D projection.

6. Speech Codec Identifier Model: Training and Testing

The codec identifier proposed in this paper needs to be trained in order to identify unknown bit stream samples as illustrated in Fig. 3. For training purposes firstly sufficient amount of chaotic type features are calculated for each codec type to form the combined feature database. Then the entire combined features database, which is composed of several chaotic features vectors obtained from the universe of speech coders, may be normalized according to their mean and variance. The parameters of the normalization process are saved and returned as an output. These normalization parameters will be used to normalize the chaotic feature vectors of suspicious bitstream samples. Finally relying on the assumption

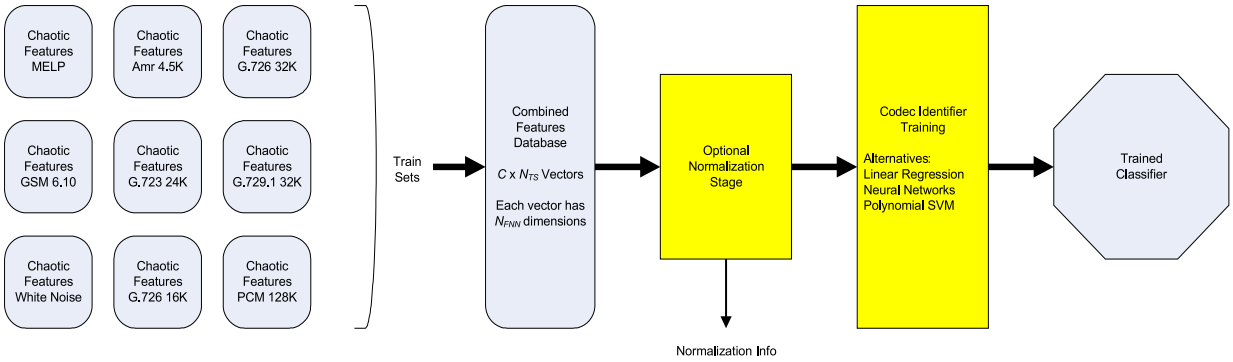

Fig. 3 Training of the codec identifier model

that different codecs would somehow have different distribution of chaotic type features, the normalized feature database is used for classifier training.

After the chaotic analysis stage, N_{FNN} features ($N_{FNN} = 15$) are calculated per sample where the combined training database is actually a two dimensional matrix with N_T rows and N_{FNN} columns. ($N_T = C \times N_{TS}$, N_{TS} is the number of samples per codec, C is the number of codecs to be identified, and N_T is the total number of vectors in the combined training database). For every vector of all codecs, mean (μ) and standard deviation (σ) values are computed and stored as shown in Equation 1 and 2 respectively.

$$\mu = \frac{1}{N_T C}\sum_{i=1}^{N_T C} v_i \tag{1}$$

$$\sigma = \sqrt{\frac{1}{N_T C - 1}\sum_{i=1}^{N_T C}(v_i - \mu)^2} \tag{2}$$

Vector elements are normalized with these stored mean and standard deviation as shown in Equation 3.

$$v_i = \frac{v_i - \mu}{\sigma} \tag{3}$$

After normalization, the combined training database is constructed. This database is used for the classifier training. Training can be defined as determination – calculation of the weights of chaotic features where the codec types can be distinguished in the "optimum" way. The classifier can be designed by using either LR (linear regression) [7], polynomial SVM (Support vector machine) [8] or neural networks [9].

Once the codec identifier has been trained, identification of an unknown bit stream sample (testing) is performed by using its optionally normalized chaotic features as illustrated in Fig. 4. If a normalization process is going to be employed, chaotic feature vector of the unknown bitstream is normalized according to Equation 3. The mean and standard deviation values are obtained from the normalization info which was calculated during training.

7. Performance of the Proposed Codec Identification Model

For all possible codec types, PCM speech files in the database are coded and partitioned into train and test databases. These databases are fed into chaotic analysis to produce test and train chaotic features data sets.

The PCM database is composed of 2560 speech samples, which are chosen from NTIMIT database [10], where each sample's duration may vary. 512 of the speech samples are used in training database, where the rest (remaining 2048 speech samples) form the test database. Note that a speech sample is either in the training or test data set.

The chaotic analysis is applied by executing TISEAN library's [11] false nearest neighbors software for five different embedding dimensions (d_e = 1, 2, 3, 4 and 5). The software produces three chaotic features per embedding dimension and therefore the concatenated feature vector of dimension 15 is calculated for each speech sample.

The codec identification model implemented in this study intends to classify 9 different types of outputs. Namely, AMR 4.5K, G.726 24K, G.726 16K, G.726 32K, G.729.1 32K, GSM 6.10 [12], the Federal and NATO standard MELP (Mixed Excitation Linear Prediction) [13] and white Gaussian noise (WGN). For classification and normalization purposes, five alternative combinations of three types of classifiers are examined as described in Table 1.

The performance of the training process is calculated according to correct identification ratio (CIR) which is defined as the ratio of total number of true positives divided by the total number of bitstream samples tested. (Equation 4)

$$CIR = \frac{\#TP}{\#TP + \#FP} \tag{4}$$

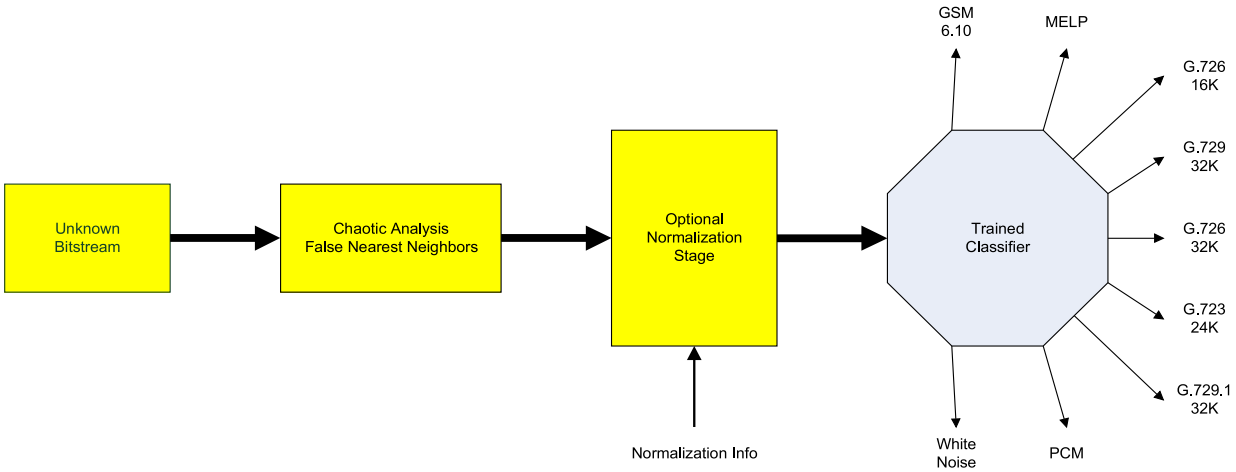

Fig. 4 Codec identification of a suspicious bit stream sample

Classification and normalization combinations used in the proposed codec identifier Table 1.

Classifier Type	Applied Input Normalization
Neural Network	No normalization
Neural Network	Variance
Linear Regression	No normalization
Polynomial SVM	No normalization
Polynomial SVM	Variance

In Table 2, test performance of the trained codec identifiers are presented. With respect to the test results, it is observed that three factors strongly impact the trained identifiers' CIRs:

Classifier Type: Since the chaotic features belonging to different type of codecs could not be properly separated by using planes, linear classifier does not offer high CIR. Non-linear SVM and neural network classifiers yield better and acceptable CIRs.

Window Size: The evaluation length chosen for the speech samples is an important parameter. As the evaluation length, which is the windowed bitstream sample, is increased the confidence of its chaotic features also increases. Unfortunately, because of some practical restrictions like sample collection time, maximum transmitted packet size and calculation complexity of chaotic features, window sizes may not be easily enlarged. Therefore only three of the window size options, 512, 1024 and 1536 are taken into consideration. 512 is chosen since neural network and SVM classifiers start to possess CIRs above 50%. 1536 is an important stopping criterion because it is both divisible by 512 and it is very close to maximum ethernet data size of 1500 and most of the classifiers exhibit better than 90% CIRs. As a matter of fact, in both training and testing cases, chaotic features which are considered as digest – hash values, can be acquired from variable length bitstream samples, training and testing are not needed to be done by taking equal sized bitstream samples as inputs.

Input Normalization: Statistical properties of the codecs influence almost every chaotic feature with varying penetration quantities. Normalization of these features as a preprocessing prior to classification stage improves the convergence rate of the training and reduces the round off errors caused from numerical calculations. Normalization improves performance in all cases.

Performance results of trained codec identifiers Table 2

Classifier Type	Applied Input Normalization	Correct identification ratios for alternative windows sizes in %		
		512	1024	1536
Neural Network	No normalization	53.3	90.5	91.8
Neural Network	Variance	56.2	91.6	95.6
Linear Regression	No normalization	18.5	24.5	34.0
Polynomial SVM	No normalization	75.0	88.5	90.3
Polynomial SVM	Variance	84.4	97.5	98.7

In Table 3, confusion matrix of the most successful codec identifier combination (Polynomial SVM trained and tested with normalized chaotic features calculated from 1536 byte length samples) is presented. AMR is the most accurately identified codec type and MELP 2400 is the least. The highest confusion is reported between MELP 2.4 Kb/s and G.726 24 Kb/s voice coders at around 2%.

Confusion matrix for the proposed speech codec identifier for eight common speech coders and white Gaussian noise bit stream Table 3.

		Claimed Identity								
Actual Codecs		AMR 4.5K	G.726 24K	G.726 32K	G.729.1 32K	G.726 16K	PCM 128K	GSM 6.10 13.2K	MELP 2400	WGN
	AMR 4.5K	1.0000	0	0	0	0	0	0	0	0
	G.726 24K	0	0.9707	0	0	0	0	0.0049	0.0244	0
	G.726 32K	0	0	0.9956	0	0	0.0029	0.0010	0	0.0005
	G.729.1 32K	0.0005	0	0	0.9990	0	0	0	0.0005	0
	G.726 16K	0.0005	0.0005	0	0.0005	0.9961	0	0.0005	0.0010	0.0010
	PCM 128K	0	0	0.0024	0	0	0.9976	0	0	0
	GSM 6.10 13.2K	0.0010	0.0151	0.0005	0	0.0020	0.0015	0.9653	0.0142	0.0005
	MELP 2400	0	0.0205	0.0005	0	0.0059	0	0.0093	0.9639	0
	GWN	0	0	0	0.0020	0.0024	0.0005	0.0005	0	0.9946

Unless inter bit relationships and bit distributions are not same, chaotic features of two different codec's outputs are expected to dissociate. But this dissociation does not always mean that, unsimilar bitstreams' features dissociate more. Training is performed by using every codecs output, and there may be many optimum solutions that depend on the training set contents used.

8. Differences between Steganalysis and Codec Identification

Although the proposed codec identification model is a derivation of the chaotic feature based steganalyzer model there are several differences, as reported below:

Ineffective feature elimination: In steganalysis existence of hidden data may change only few of the features significantly (in other words most of the features may nearly be orthogonal to secret data existence) as a result of this circumstance, for the purpose of reaching maximum decision accuracy impotent features should be discarded. For codec identification however, statistical properties of the codec influence more chaotic features and therefore there is no need for elimination stages.

Feature types: Since existence of secret data does not reveal itself in every chaotic feature in a distinctive manner, the performance of the steganalysis gets better as the number of investigated chaotic features is increased. Consequently chaotic features are the collection of false nearest neighbor results and Lyapunov coefficients in steganalysis. The test results however tell us that usage of false nearest neighbors is sufficient for an acceptable CIR in the codec identification case.

Non linear classifiers: As steganalyzers possess much lower CIRs than codec identifiers, usage of non linear classifiers in steganalysis may not enhance CIR and may force the steganalyzer to memorize the applied training sets. Conversely, with respect to test results presented in this paper codec identifiers with non linear classifiers have significantly higher CIRs.

9. Conclusion

In this study a codec identification model, which identifies the codec type being used, is proposed. The identification model is derived from a steganalyzer model proposed by Kocal et al. [1], where decisions are based on chaotic features calculated from suspicious bit stream samples. The identification model is examined in two steps. Firstly training is described, and then test results and performance limits follow.

In the training, a chaotic feature database is produced from coded speech samples which hosts adequate number of bit stream samples belonging to different speech coding algorithms to be identified. From the assumption of different codecs produce outputs with different statistical properties, calculated chaotic features are shown to be distinct according to the codec being used. From this chaotic feature database, weights which optimize the separation are computed for the classifier. Certainly, the term "weights" has different meanings for different classifier types; in the linear regression case, weights mean a vector, while it means the excitement ratios of all the inputs of all the neurons in a neural network.

During the test, the trained codec identification model is tested using a number of coded speech samples. For each coded speech sample, initially chaotic features are calculated; then these calculated features are fed into the trained codec identifier where decision correct identification ratio of identification model is checked. According to the test results, SVM based codec identifiers accomplish better than 97% correct identification ratio for 1024 and 1536 byte size windows for nine different outputs. Similarly neural network based identifiers achieve above 90% success rate. On the other hand, codec identifiers based on linear regression could only achieve correct identification ratios between 18-34%.

In summary, the proposed codec identification model proves that the type of a speech coder used in a unknown bitstream can be revealed by using chaotic features. Although the proof of concept is carried out on speech, similar approach can be adapted to audio or video which will improve network monitoring applications.

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Jakub Soviar – Josef Vodak *

VALUE NETWORK AS PART OF NEW TRENDS IN COMMUNICATION

This paper deals with the topic of value network importance for the new trends in communication. It describes the main definition of distribution and place, and their relations with the value network. The value network is described as a network of all relevant factors impacting the product where the product is in the network's centre. The case of Apple iPod is provided as an example of the value network. E-commerce and digital distribution are also described in the paper as new communication trends. Discussion deals with main text results and authors' opinions.

Keywords: *marketing, management, value network, distribution, communication, new trends, innovations, point of sales, digital distribution, e-commerce.*

1. Introduction

In the time of globalisation, internetisation, social networks, information spreading and new way communication the ability to make up something new, creative, efficient and unique is a key factor of success. It is something which can bring higher value for customers and other stakeholders. The chance to come to the market with an idea which nobody has ever had yet, nobody has had it patented or even nobody has run business with it seems to get smaller and smaller. Just open the Internet and the idea which seemed to be a discovery may suddenly lose its potential rapidly in global competition.

Innovations have an essential role in the current economic, political and social transformation of developing companies and countries. We can look at innovations from a few points of view. We can talk about the product innovation (including the products and services the specification or intended use of which differs from previous company products significantly), process innovations (occurring in industry as well as in services and including the new or improved production methods, logistics, supplier and distribution systems), organisation innovations (implementation of important changes in business practice, organization of working duties and a process of making decisions), marketing innovations (important changes in the way of trading with products and services including design and package changes) [1, 2, 3].

A complex view which connects all above mentioned kinds of innovations is important here. It is not easy to innovate at all as shown by the results gained by the authors in cooperation with the entrepreneurship practice as the co-providers of the project Innovations – the Way to Increase Competitive Ability and Regional Development (cross-border cooperation program Slovak Republic – Czech Republic 2007 – 2013, ITMS 22410420011). Therefore we want to discuss value networks in this paper and we want to point at some examples from abroad how innovatively some well-known foreign companies have dealt with this issue and to inspire a lot of our companies. In the time when the examples appeared, they embodied interesting and innovative solutions.

Introduction to the topic

A traditional definition describes distribution as a complex of activities whose goal is to make a finished product available for customers. The distribution comprises [4, 5, 6, 7]:

- Processes of physical relocation – the purpose of distribution is to transfer a product in the right time to the right place. Physical distribution includes transportation, storage and inventory management;
- Changes in proprietorship – the change of proprietorship of the product occurs at least once in a distribution process (in case of so called direct sale, i.e. the sale performed directly between a producer and a final consumer, to compare with the indirect sale, in which a product goes through several processes of the sales and purchase on the way to a consumer: e.g. whole-sale companies – retail companies – consumer);
- Other activities – we mean e.g. gathering of marketing information, advertisement, insurance, loaning etc. These are the activities which are not involved in the previous processes directly; however, they create conditions for their smooth running.

Such activities are performed either by the producer or by different organizations which constitute so-called *distribution net-*

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work. The distribution network forms a framework to perform the functions such as purchase, sales, storage, transportation, goods movement financing, risk resistance, marketing information providing. The distribution network consists of the following independent participants such as producers, distributors, sales organizations (whole-sale and retail organizations) and other organizations intended to perform specialized (supporting) activities [8, 9, 10, 6, 7].

According to the above mentioned we can say that successful business (marketing) process depends on a lot of factors. One of the most important factors, after the value creation for a customer, is its delivery. [11] In general we distinguish the following ones [10, 12, 6]:

- Value network: a complex process which includes necessary information inputs, inputs for the transformation process, partnership relations, delivery of the value to the customer, feedback etc.
- Marketing distribution channels (complex logistics chain): particular elements of distribution network, their quantity, quality and management (technology, processes, people, organization, operation ...).
- Parameters of a place where a customer meets the product: physical attributes of the place (architecture and geographical location), education and behaviour of the personnel, parameters and possibilities of the web site etc.

2. Distribution Channels and Value Networks

"Marketing distribution channels are a set of mutually dependent organizations engaged in the process of making a product or service available for use or consumption from the formal point of view. It is a set of ways of releasing a product or service after having been made and which finishes by their purchase and use by the final user" [10, 12, 6].

Every company should make a strategic decision on the *importance of particular* channels. That means to specify their importance and thus the rate and intensity of their management. Utilisation of the complex channels can decrease costs and vice versa. The same applies for a direct channel. According to relevant literature [10, 12, 6] the participants of branched distribution channels earn approx. 30 – 50 percent of the final purchase price of a product [6].

The process of the *DELL* company is a good example of reassessment of the importance of the marketing distribution. The company was established in 1984 as a "student IT company". Their business model was built on direct distribution. They used catalogues. Orders were made by post and later on by telephone. In that time that was an important innovation. A customer built a computer and accessories himself. The order was delivered where he had stated. Already in that time DELL started to develop their unique service and guarantee conditions. That completed the model of direct distribution and significantly affected customers' satisfaction positively. DELL started the sales via the internet in 1996 which enabled even higher comfort when configuring the orders and their updates. However, the development on the market caused that the competitors of the company DELL, who used distribution through the chain of brokers started to increase their market shares and profit at the expense of DELL. That was caused especially by:

- Increase in shopping in shopping centres and chains. Customers buy the products which are available on the places where they move most often.
- Possibility to show the product next to the competition. In retail shops the products are next to each other. A customer may choose immediately according to his preferences.
- Cost saving. Direct distribution may have extra high costs. When distributing through a chain of brokers it is possible to decrease a big part of such costs. E.g. personnel costs, database administration costs etc.
- Access to new/other customers. The distribution through the chain of brokers brings the product closer to customers. It brings it to the places where the customers, who would not be affected by the distribution channel, move.

The above mentioned facts results in the state that the competition and market development have changed the situation significantly. DELL had to respond to it by the change of the distribution channels importance. Nowadays they are operating a direct channel as well as distribution through a chain of brokers.

Within the distribution channels we talk also about *their development.* It is establishing, development and leaving the channels regarding to company goals and market development. That means their complex management. The example of DELL relates also to that phenomenon. A common example is a "brick and mortar" computer shop in a Slovak or Czech town. It is a communication and distribution channel for the customers from that town. If it makes an e-shop, another distribution channel, it can cover the national market, gain more customers, get closer to the competition etc.

Another real example of the *distribution channels development* is the process of the *Coca Cola* company. It is a global corporation which has already expanded to the whole world. At such expansion it must use and work in local conditions and use local resources (inputs, people...). Therefore they build local production and distribution centres in strategic locations which are able to cover the given market with their production and distribution. Coca Cola themselves have specified the target destinations of their products, i.e. the places where they desire to sell their products to target customers:

1. Retail food and mixed goods shops (super and hyper markets, small shop unions etc.)
2. Drink machines (located in schools, public areas, waiting rooms, in different companies etc.)
3. Hotels, restaurants, cafes, including "fast food" etc.
4. Mobile sale (any mobile sale where drinks may be offered or solely drinks are offered)
5. Amusement parks (water parks, theme parks such as Disneyworld etc.)

2.1 Value Networks

In its wider meaning the company is in the centre of the value network. It forms a system of partnerships and alliances concluded by the company in order to supply and deliver its offer. The value network includes company suppliers and their suppliers, also company direct customers and final customers. The value network also includes value relationships with other essential entities (stakeholders) such as research centres, particular researchers at universities or governmental inspection bodies [10, 12, 6, 13, 14].

It is a relatively new concept built on a complex system approach. It deals with all relevant events which affect the process of value creation and delivery. The example of the product of the company Apple, a well-known and popular iPod, may serve as a good example. It is a simplified model of the value network shown in the following Fig. 1.

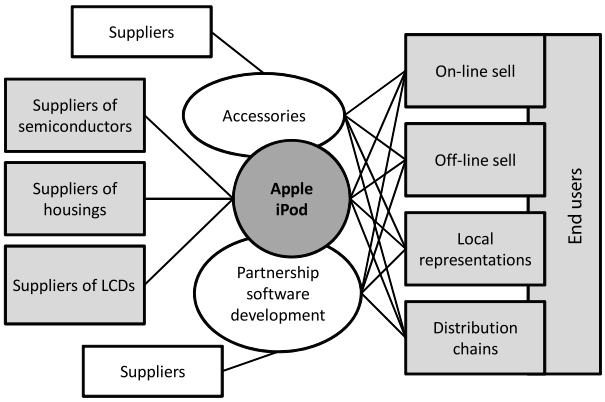

Fig. 1 Example of a value network (own solution)

The product, iPod, is in the centre of the value network. On the left there are suppliers of the main parts needed for physical production of the product (suppliers of semiconductors, covers and LCD displays). In the centre there are, together with the product, partners who must cooperate with the company closely, since their products and connected with iPod directly. They are the producers of original accessories. If this is to be licensed by the Apple company, it has to meet specified technical standards. The software producers must also cooperate with the company closely. Their applications must go through testing and must be certified. In both cases there is an advantage that the products (accessories and software), after having met the conditions, will be included in the official Apple distribution network. Here we have gotten to the distribution ways of the product shown on the right. In case of iPod that can mean the official channels of the Apple company or their partners. The product is available also in free distribution which means they can be the distribution channels completely independent from Apple. Each above mentioned distribution system is interconnected with particular final customers. Apple is thus able to satisfy the demand in a global rate. Feedback has not been shown in the figure. Of course, there is feedback. It is the most intensive between directly interconnected customers and producers of the accessories and software used by the customers.

2.2 Design and Principles of Distribution Channels Management

Let's look at the basic categories of the decisions which must be made by the management regarding to the *marketing channels design.* They are the categories which will be filled with particular contents in a given case [10, 12, 6]:

1. The analysis of the levels of the provided services required by the customers. The analysis focuses on the amount, waiting and shipping period, territorial coverage, product diversity and additional services
2. Setting the goals and limitations. On the basis of the results of the analysis exact goals regarding to the distribution are set. Limitations, e.g. financial, personnel, quantity ones can be stated, too. In fact they further complete and specify the stated goals.
3. Identification of main variants of the distribution channels
   - Types of brokers – particular brokers who will perform the work in particular distribution channels (people and organizations)
   - Number of brokers
     - Exclusive distribution – used in the case when the producer wants to keep intensive control of his product. It is represented by strict and exact limitation of the number of brokers. The producer defines who can distribute his products. It is often used with exclusive goods and luxurious brands (e.g. Gucci, Versace, Lamborghini etc.)
     - Selective distribution – only a few brokers are used, not all of those who want to distribute or sell the product. The producer may define the criteria. Middle rate of control.
     - Intensive distribution – the producer wants to get his product into as many shops as possible. The lowest rate of control. It is used especially at common consumer goods such as cosmetics, sweets etc.
   - Conditions and obligations of the distribution channel members. The producer can specify them and check if they are observed. The possibilities are derived from the producer's prestige and economic power.

4. Evaluation of the main alternatives. It is proper to prepare a few variants of the distribution system. Then those ones which can best fit to the specified goals and possibilities may be chosen from the available possibilities.
   - Economic criteria related particularly to the efficiency.
   - Rate of control and adaptability, i.e. what rate of control of the distribution channels the producer wants to have; what are the possibilities of the distributors to adjust to new or modified conditions (e.g. new products, new technical documentation, modified warranty conditions, etc.).

2.3 Point of Sales

In this case we are talking about the place where a customer meets the product. Usually the physical attributes of the area are concerned, i.e. the architectonic solution of the exterior, interior, as well as its geographic location. In its wider meaning we can talk about the personnel offering the product parameters, i.e. their abilities, knowledge, behaviour, clothes, appearance etc [4, 8, 9, 10, 12, 6, 13, 7, 14]. Nowadays it is of course also the level of the web sites where the products can be ordered or where the information about them can be found (limpidity, information, possibility to order and pay, possibility to provide feedback, graphics, design etc.) [5]. A complex view therefore takes physical, design, technical and personnel parameters into consideration – their level, quality and quantity.

The sales point in the direct control of the Apple company can serve as a good example. They support, by all their parameters, the image and philosophy of the company in form of care for the Corporate Identity. The company premises are situated in city centres, they have modern architectonical solution corresponding to design principles of the company, the staff is highly motivated and their relationship to the company products in very close.

The similar example from a different industry is represented by official branded car services of the Toyota company. The company has specified a few principles the satisfying of which is a matter of course. The walls and floors are shiny white and the whole premises are maintained extra clean. This has a positive communication effect on a customer. Connection with cleanness and preciseness is made here.

Upon the above mentioned facts we can state that the parameters of the place where a customer meets the product have an essential effect on the internal and external communication of the organization [15]. Such parameters affect especially the image (of the product, brand, company, operation...), position of the brand (regarding to the competition, customers), satisfaction of a customer (potential and measured) or self-specification regarding to the competition (especially differentiation).

2.4 E-commerce and Digital Distribution as Current Trends

Information and communication technologies and their development change the market substantially. Usually very rapid changes are concerned. There is a new, non-physical value in form of digital contents there. It can have a form of the data, software, music, video, games etc. the word "multimedia" is often used in this meaning. New innovations from new technologies, methods and opportunities can bring significant benefits to an enterprise [16]. Regarding to the distribution we talk particularly about electronic trade and electronic distribution.

The following are defined as basic terms related to this topic: E-business (designates utilisation of the electronic means and platforms for the purpose of business), E-commerce (a company or a web site offers or sells products on-line), E-purchasing (a company purchases goods, services and information from different suppliers on-line) and E-marketing (a company informs purchasers, communicates, promotes and sells their products on the Internet) [6].

The business model of the *Amazon* company can serve as a good example of *e-commerce.* It started in 1995 as a so called "pure-click" company, that means the company focused on solely on-line ordering system through their web site. Originally it was a book shop. Step by step they have extended their range of goods. Nowadays they are selling and distributing almost all portfolios of consumer goods (e.g. clothes, electronics, cosmetics...). Their competitive advantage was their sole focus on the Internet. Their sales point is their web site. That was just here where they have brought the following essential innovations:

- As the first ones they made it possible to *comment and evaluate* the products being sold. A user had to register. Everyone could see the evaluation. They started to make the product charts according to the users' evaluations. The customer thus got bigger trust to a then new form of shopping. At the same time they attracted new customers who were looking for some information regarding to the use of some products etc.
- The company started to *gather the data* on their customers. Since the registration is required, they monitored the facts such as what the person looks for, what he comments, how he comments, how often he buys, what amount he spends etc. Upon such data Amazon adjusts the offer, customizes the offer for the particular person.
- Nowadays Amazon has expanded also in the digital distribution of music, games and electronic books (described further in the text).

Digital distribution is an intensive trend of these days. Technical possibilities of communication networks are still getting bigger. The development since 2000 in the Slovak Republic can serve as a good example. Internet connection was then made mostly using the Dial-up technology with the theoretical highest transmission speed of 56kb/s. Nowadays mobile and optical networks are common. In 2010 the average connection speed was 10, 18 Mbit/s in Slovakia [17]. The price of such connection is the same as for Dial-up in 2000, often even lower. The data storage places such as hard disks have undergone the similar development. Their capacity has been increased significantly and their price has been decreasing permanently. In 2000 common hard disks had the capacity of approx. 1 GB. The capacity of the current hard disks is normally in the range of terabytes.

Videogames, music, films, software and data form digital contents. As it has been mentioned above, the technical capacities today enable their on-line transmission. We can see here a sudden regression of physical media (CD, DVD etc.) and sudden launching of on-line distribution, cloud computing, video on demand etc. Mobile devices participate in this trend and support it. A mobile phone has not been only a communication device now. It is a centre of personal joy and planning. The contents distributed through the communication networks to final users is a product – it makes value.

Without applications, or music and video such technologies are irrelevant. The demand in this area tends to rise, especially in the area of digital joy.

The products of the *Apple* company are a good example of utilisation of the digital distribution as a competitive advantage. Particularly the products iPod (digital mobile music player) and *iTunes* (software for administration and purchase of music and video) are concerned. Apple noticed enormous users' demand for music distributed through the Internet at first in the MP3 format. However, such demand was satisfied illegally. Therefore Apple created a legal solution. It offered a complex product in form of a mobile player and a software, which enabled buying music onlone simply and legally. The products were interconnected regarding to the function – they supported each other. E.g. the first generation of the players, launched on 23/10/2001, was equipped with the FireWire interface, which is the Apple's patent and a competitor of USB. FireWire was used in the computer platform MAC, at PC this interface was rather an exception. Of course there was a fact that the player required the cooperation with the iTunes software for its proper function, e.g. when transferring music from a computer to the player etc. The products were made in accordance with the philosophy of Apple. There was an emphasis on the design of the player and function of the whole solution. This was really high – however, it required the using of the whole platform from Apple. We can say that Apple satisfied a real need on the market in form of the first commercial system of the digital music distribution. They became market leaders and they have not lost this position so far.

Another current and suitable example is a *digital distribution of books of the Amazon company*. Amazon sells cheap, simple electronic book reader based on the e-ink technology. This technology simulates real paper and is extra energy saving. The reader is supported by the on-line sales of electronic books. The system is comfortable for the users and enables to send the purchased titles directly to the particular reader to the particular user. We can say that Amazon was inspired by the process of Apple in case of iPod and iTunes. However, their solution is more open. The reader supports standardized formats and no special software is needed to download books to the reader. Amazon even provides free text conversion to the needed format on-line and free of charge. Let's look at some data regarding to the market success of this model [18]:

- In 2010 Amazon sold 143 electronic books for the Kindle reader for each 100 classical books.
- Net income from sales in 2010 achieved USD 12.95 billion, which means the inter-annual increase by 36 % to compare with the fourth quarter 2009.
- Net profit increased by 8 percent to USD 416 million inter-annually (2009 – 2010).
- Income from sale for the whole calendar year 2010 reached USD 34.20 billion.
- Net profit was by 28 percent higher inter-annually (2009 – 2010) and reached the level of USD 1.15 billion.

3. Discussion

The development of the information-communication technology (ICT) has changed the whole society. The distribution processes are not an exception. The traditional attributes have not lost the importance, however, the significant influence of ICT is obvious e.g. at placing orders, monitoring of the way of the goods, warehouse management, logistics etc. Produced values with significant demand have also moved to the digital world. "Proper use of ICT can significantly affect the creation of business strategy [19]." The data which we make are often more valuable than the hardware we work with etc.

The above facts result in a few basic recommendations related to the current issues of the value distribution:

1. Focus on stakeholders, i.e. the entities which are essential for the organization and able to affect it significantly, is important [20, 13]. The distribution chain, sellers, web sites administrators, companies providing the hosting and functioning of the web site or orders... these all are important stakeholders. Therefore, it is necessary to specify, clearly and exactly, who are our stakeholders and what value they expect.
2. The analysis of the possibilities to make a base and initial point for distribution ways planning. Today's market provides a lot of ways to distribute the value more efficiently or to target more customers or to target a specific group of selected customers etc.
3. A complex view to the production and consumption of the values comes from the system approach. The value network is a suitable help to understand the importance of individual relationships, communication and their subsequent control.
4. Particular distribution channels have different importance and different value. According to these attributes, which should be known at first, it is then necessary to approach the management of the relations within the given channels.
5. Physical attributes of the place where a customer meets the product affect his decision and satisfaction.
6. The Internet and mobile communication are a matter of course today. At least basic Internet support is suitable for almost every company.
7. Web site is a communication platform. It represents the corporate and product image. Its quality and services which it provides affect the customers' decision and satisfaction.
8. The Internet and mobile communication represent a possibility how to expand, how to get more customers.
9. Internet trade and digital distribution are strong trends. Digital contents is very attractive for people. It is important to care about the current, comfortable and safe possibilities of its distribution to the customers.
10. The distribution does not refer only to the product itself but also the corporate values and Corporate Identity. Therefore, we cannot forget that the additional services to the products such as consulting, information, warranty, complaints, installation, trainings etc. are concerned, too. This all can mean a significant competitive advantage.

We should realize that there is not a single clearly best way of distribution. Combination of them is often suitable, e.g. a direct channel, a chain of brokers and Internet support. The fact that the only guarantee is a guarantee of change applies, too and the same applies also for the communication with a customer and distribution of value for a customer and for stakeholders. We could realize such changes at the above mentioned examples of the global companies.

4. Conclusion

We have discussed the distribution, value networks and utilisation of the information communication technology in innovative approaches to communication and value supply for stakeholders in the papers. The article was intended to stimulate the concerned to think about the fact how it is possible to modify the current approaches of companies creatively into a new quality in form of the examples.

The value for a customer will not be a value which he requires if the distribution does not contribute its share. The focus on the idea of the value network may be interesting for a lot of our companies (therefore also this article), on the other hand we should realize that creative and innovative approach is necessary and very important also in this case.

Unfortunately a lot of our companies cannot see anything in our ideas presented above which we should deal with intensively today. It is often caused by the fact that a lot of them have their energy focused on many operative activities and they have not created enough space for strategic thinking e.g. in form of the study of the best practices to the distribution and value networks.

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Slobodan Ristic – Zoran Cekerevac *

DEVELOPMENT OF LEADERSHIP COMPETENCE AND SELF IMPROVEMENT OF THE LEADERS

To be a good leader one always has to have a vision. He knows what he wants to reach and how to get it. In such competitive environment as it is today, leader must prepare himself to daily new challenges. This article deals with leadership, and especially with leader's self improvement and development of leadership competence. Introductory part of the article gives one of possible classifications of leader's skills with a brief explanation of each named skill. There are also given some experiences of some of the most successful world's companies in the area of the strategic rotating and moving managers to appropriate leader's positions. In the article, there are also analyzed contemporary trends in development of leadership skills and the five phases of development of leadership development according to the Witten-Cameron model. Under Conclusions, there are summarized results of the analysis.

Keywords: *Leadership competences, Witten-Cameron model, leader's skills, Kurt Lewin's model, Kolb's model*

1. Introduction

The strategic studies that exist for about 25 years have shown that the biggest cause of stress at workplace is incompetent managing. These and other relevant data indicate the conclusion that the effective leadership is the primary factor of organizational success [1].

In available literature it is almost not possible to find examples of classification of human's skills. This is the consequence of extreme complexity and uncertainty of this phenomenon. After all, some classifications are possible. If here is applied the established model that exists for skills typology then we could put all the skills into four groups:

- Intellectual skills
- Sensory skills
- Motor skills and
- Sensory-motor skills

In context of organizational behavior, intellectual skills are the most important. Those could be cognitive skills related to a human intellect and the power of thinking and judging, but those skills are almost always seen as abilities [2]. Often, they can be divided on managers, leadership, negotiation, sales, social skills, etc.

Sensory skills are hardly ever mentioned, because the perception is categorized as the group of abilities connected to human's senses. However, and the skills of vine and food testing (enological skills), the interpretation skills of sensor notes (work on radar, sonar, scanner...), the details observation skill, etc. could be put into this group.

Motor skills are often mentioned, especially in psychology and pedagogy. These skills are related to mobility and dexterity of human's body. Most of the researches that are done so far concerned motor skills. One of the first surveys of motor skills, the Time Study, was conducted by Frederick Winslow Taylor in the first decades of previous century. Later researchers, starting with Frank and Lillian Gilbreth, gave focus on the motion study, complex phenomenon like assembling and disassembling small arms, typing skills, playing instrument, dancing, etc. All those skills can be described as the organized compositions of movements and actions. Today, the both studies together they are called "a time and motion study" and represent a business efficiency technique [3].

As the combination of sensor and motor skills is possible, it is also possible to make the group of sensors-motor skills [4].

The differences that can be seen these days are connected to usage of different learning methods. The classical leadership training referring to classroom, now is largely completed or even completely replaced with other activities, such as lifelong learning, training, courses, mentoring, evaluation, etc. On the other hand, more attention is paid to emotional alignment with followers of the leaders. More precisely, the reason is that in contemporary conditions only the absolutely committed employees become the key to success for the entire organization. The greatest attention is given neither to what the leader is, nor to abilities and skills that he possesses, but to the process which is the result of the interaction between the leader and the followers.

Contemporary trends in development of leadership skills have two directions [5]:
- Multiplication of methods of developing leadership skills and

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- Emphasis on the importance of emotional leadership characteristics.

There are numerous models and methods of leadership skills improvement. One of the most accepted models of development those skills are the model proposed by David Witten and Kim Cameron. According to them, development of leadership skills is a process that is cyclically carried out in five phases, namely [1]:
- Evaluation of knowledge and skills,
- Collection of knowledge about effective practices for improving knowledge and skills,
- Case studies,
- Practicing the newly acquired knowledge and skills to cases, and
- Application of learned knowledge and skills in personal work environment.

In the Table 1 there are presented the contents and goals of each of these five phases. One can see now, that the essence of the development of leadership skills makes, at the first, the willingness of an individual to go through changes, then permanent learning, detailed analysis of case studies, adopting and practicing of the guidelines of behavior, building the personal leadership style and finally applying knowledge in real terms and, of course continued work on it.

The business environment in which companies operate today is much more turbulent and complex and therefore less predictable.

Contemporary organizations have to show exceptional dose of flexibility in order to survive and succeed in demanding market. Thus, both increasing at need for leadership skills and abilities, and techniques are changing their promotion and development. Today, all practical methods of development of leadership skills could be classified into two main categories:
- Leadership training, and
- Leadership development activities.

2. Self improvement of the leaders

In the modern business world a number of programs and trainings for developing leadership skills are developed. Each leading company or SME is trying to give its contribution. There are, also, a significant number of educational organizations with accredited programs for leadership competence development.

The key feature of the majority of leadership development programs is that they are mainly focused on the skills, and less on knowledge and very rarely on abilities and personal qualities of holders of this business and social function [5]. There is no doubt that such programs are necessary and useful but not sufficient for a general leadership development. In particular, the presence of many development and training programs does not promise much to a young man seeking professional recognition in this field and that passes through it he will become a leader. In this sense, Bennis and Goldsmith [6] advocate the theory that a man can become a leader only by discovering his natural energy and desire and with persistent effort to find his own way to use energy and achieve desired goals. Educational programs and trainings continue to play a role of drilling natural crystals. “Many young people with significant innate talent for leadership often do not to achieve what is in them. So the part of our task is to develop what is naturally Leadership in us, but this requires cultivation. Talent is one thing and another is winning demonstration. Generally speaking, the developments of any succeed complex talent searches happy combination of motivation, character and circumstances. Most of the human talent remains undeveloped.” [7]

Like the gardener who is not a plant finder, and who is not even a creator, but just a raiser, so the leader is not born as a leader, and is not created. He is educated and trained to. From most previous studies and further, based on the analysis of their own experience of respected individuals, it can be concluded that the leadership can be learned. But that is not all that is required; it is also required the basis on which the great knowledge will be built. So, leaders are neither born, nor made. Instead of that, they shape their inherent potentials into experience, enabling them to develop skills needed to solve major organizational problems. There is a significant relationship between the emotional maturities of managers, which manifests itself through the awareness of their own personality and empathy, and their financial success. “Good people”, people who understand themselves but also others around them, first complete the task. Being skilled means to be able to deal with the diversities within the organization. It is a skill to take the best from a variety

Phases of cyclical model of development of leadership skills — Table 1

Component	Content	Goals
Rating Skills	Instruments for measuring, Playing roles	Evaluate the current level of knowledge, skills and abilities and create readiness for change.
Learning Skills	Written text Behaving Guidelines	Learn correct principles and present the explanation of guidelines for behavior.
Skills Analyze	Cases	Provide examples of appropriate and inadequate performance terms of skills. Analyze principles of behavior and the reasons of their action.
Practicing Skills	Practice, simulations, role-playing skills	Practice guidelines for behavior. Adopting the principles for personal style.
Application of Skills	Work Tasks (literate through behavior)	Learned in the classroom transfer to the actual work station. Encourage continuous personal development.

of experiences, skills and perceptions of others with whom one cooperates. The Skill of establishing relationship with other people is in the very own heart of leadership. The criteria for measuring success in leadership are not based only on gained knowledge and skills but also on how those who work in this area can successfully solve problems, how they are successful in relationships with others and how to cope with themselves. This is driven by personal qualities, such as perseverance, self-control, and ability to understand others. However, to become a leader is not easy, but the scientists said, much easier than most people think. [8] Success in solving the problems of leadership depends on the complex series of competencies that make available and necessary knowledge, skills but, also, of a personal potential. In general, the skills needed to solve organizational problems involving the creative arts solving various problems associated with problem's identification, understanding the problem and generating potential solutions. These are the social skills of reasoning, spotting potential solutions, providing a framework and application of social skills related to motivating and directing others in the implementation of solutions. In this sense, several steps in the process of leaders' self-building can be distinguished:

- Master the leadership competencies based on their own experience,
- Get rid of the myths and stereotypes about leadership,
- Understand the treatment of leadership in their organization,
- Self understanding,
- Change the way of learning and
- Learn to put up with failure

2.1 Own experience

Acquiring leadership competencies based on personal experience involve the use of all experiences, in families, peer groups, schools, and those obtained in meetings with other leaders and their own professional and leadership experience. Even the experience of young people is a great treasure of information, values and incentives. This is not a direct copy of already experienced, but the mental processing and sublimation of often unrelated experiences. Unconscious behavior patterns and perceptions created on experiences in large measure determine the current and future actions. Using own experiences in the development of leadership competencies is not the application of methods of learning through trial and failure. It is a much more complex management of personal development, based on the application proceedings "reflective practice". "Reflective Practice" or "reflection" is one of the most useful models of learning in adults. This model was designed by David A. Kolb [9] and based on theoretical assumptions about Kurt Lewin's model of learning. This model has stood the test of time, and today, reflective practice is used in many programs, in particular those whose goal is developing of leadership potential. Kolb's model of reflective practice provides four phases in the learning cycle. With Kolb's model is much easier and methodologically arranged the technique of reflective practice, and learning from personal experience.

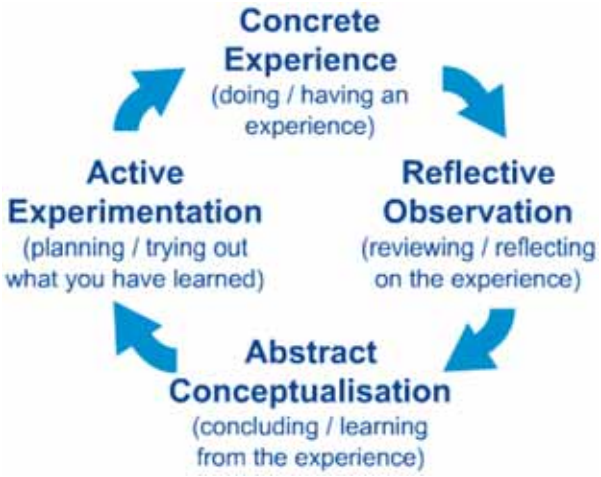

Fig. 1 Kolb's learning model [10]

According to [10], Kolb's learning cycle phases are presented in Fig. 1, where are:

- Concrete Experience – the specific event – the working arrangement – gained experience; the "Concrete Experience" is the "doing" component which derives from the content and process of learning;
- Reflective Observation – reviewing – thinking (reflection) about the experience; The "Reflective Observation" element stems from leader's analysis and judgments of events and the discussion about the learning and teaching that leader engage in with its partners, colleagues and fellows;
- Abstract Conceptualization – the conclusions – learning from experience; it allows leader to come to conclusions about the practice;
- Active Experimentation – planning – checking what has been learned. The conclusions one formed from his "Abstract Conceptualisation" stage then form the basis by which he can plan changes – "Active Experimentation". "Active Experimentation" then starts the cycle again when he implement those changes

2.2 Myths and stereotypes

Under the leadership potential for self-improvement it is necessary to get rid of the myths and stereotypes about leadership. On the way of adoption competencies that are needed to an effective leader, it is good to get rid of preconceived notions as soon as possible. The most common myths and stereotypes associated with leadership are:

- "Leadership is a very rare skill" is one of the most common myths which, of course, is not true. It is true that charismatic leaders are rare. It is true that great actors, musicians, painters are rarely born but the fact is that in each person lies some leadership potential. In various aspects of people lives exists immense ocean of leadership roles. It happens that someone is a great leader in his organization, but in any other environment, even in his own family, has a very minor role;

– "Leaders are born as such, and leadership cannot be learned" is the enduring prejudice that dates back to the first theory of leadership. Sometimes from the biography analysis of some outstanding leaders it might be concluded that their leadership role was destined from their childhood. From the biography analysis of most leaders with significant results achieved, such a predisposition or a genetic basis cannot be seen. One can conclude that some basic competencies of leadership can be learned if there is a strong motivation in that direction.

Stereotype "charismatic leaders" can be partially proved correct. There are numerous examples that testify to the charisma of individual, successful leaders but the fact for the majority of leaders is that they were such ordinary people who were not much different from peers and colleagues.

"Leaders are only at the highest levels of the organization" is another very common myth about leadership. This myth originated from the practice of leadership training and guidance to the attention of the entire top management companies. With a more detailed analysis of the distribution of leaders, the presence of a number of leadership roles and the holders of these roles can be observed.

2.3 Understanding of the treatment of leadership

Big companies are creating organized and planned more leadership roles throughout self-teams, divided decision making and creating strategic business units that operate as independent business entities.

Understanding leadership in their own organization can be beneficial for the development of leadership competencies. The goal of this effort is to look at the way working environment stimulates the expression of leadership and how and why it discourages the development of the same. This primarily reflects the explicit rewards, recognition and promotion, but in many subtle ways – through training programs, social patterns, values, communication systems and so on. [1]

2.4 Self-understanding

The next phase of self-development of leadership skills is a self- understanding and is derived from Socrates' commandment "know you (yourself)." This exercise in self-improvement business leadership can have the following working steps:

– Self-reflection, thinking of particularly critical events that had successfully or unsuccessfully completed;
– Getting to know yourself through constant contact and socializing with other people or getting feedback from the outside world on the impression given to other people;
– Openness to new experiences and knowledge, a review of earlier beliefs and attitudes in dealing with new information that are contrary to earlier beliefs related to personal priorities and goals;
– Consistency in the sense that there is a tendency of constant correlation between beliefs and behavior.

2.5 Way of learning

Acceptance of a different type of learning involves an active, self-conscious and committed learning and taking risks for fulfillment of leadership requests in today's turbulent and very complex environment. This study simply does not show the acceptance of a set of knowledge or scientific disciplines, but seeing the world as it is and what could be, understanding the differences between the two conditions and making efforts to overcome differences.

2.6 Learn to put up with failure

Finally, leader must learn how to endure a failure. The way in which, the effective leaders deal with failure is perhaps their most important quality. Great leaders do not think about it, many of them, a mistake is simply one way of doing things. Huge number of people perceived a failure as something definite and reacts to it with helpless discouragement; for successful leaders – the failure is just the beginning.

In a study by David Day carried out at 350 companies having leadership development programs and leadership skills, it is written that the most successful programs that are applied are mentoring, active learning and assessment methods. He found that two people participating in the identical 360-degree feedback experience are likely to take away different things from the experience [11]. In self improvement of the leaders a number of different procedures are applied for the same purpose.

One of them is called self-determined learning that can represent the essence of the development of leadership skills. This model of training was designed to develop or strengthen some aspect of personality, or achievement of a role model or both. This training begins with an imagination of a strong picture of the ideal I, and as an accurate picture of what we are now, present objective and observed me. This self directed learning is most effective and gives lasting results, provided to understand the change process and steps needed to it accomplished. Model of learning through personal training was developed by one of the leading researchers and practitioners of leadership – consultant Richard Boyatzis.

According to the systematization of process of self-determined leaders' learning, it takes place in five phases or, as the authors mean, "Discoveries". Their goal is that a man through the changes becomes an emotionally intelligence leader who possesses the necessary competencies. This type of learning requires repetition, the steps are not carried out smoothly and evenly, but each of them requires a different time and effort. By practicing, new habits over time become part of the actual repertoire of behaviors. Changing habits, improving of emotional intelligence and leadership style, lead to desired aspirations and getting closer to the goal, achieving the "ideal I". The cycle of self-directed learning is a lifelong process,

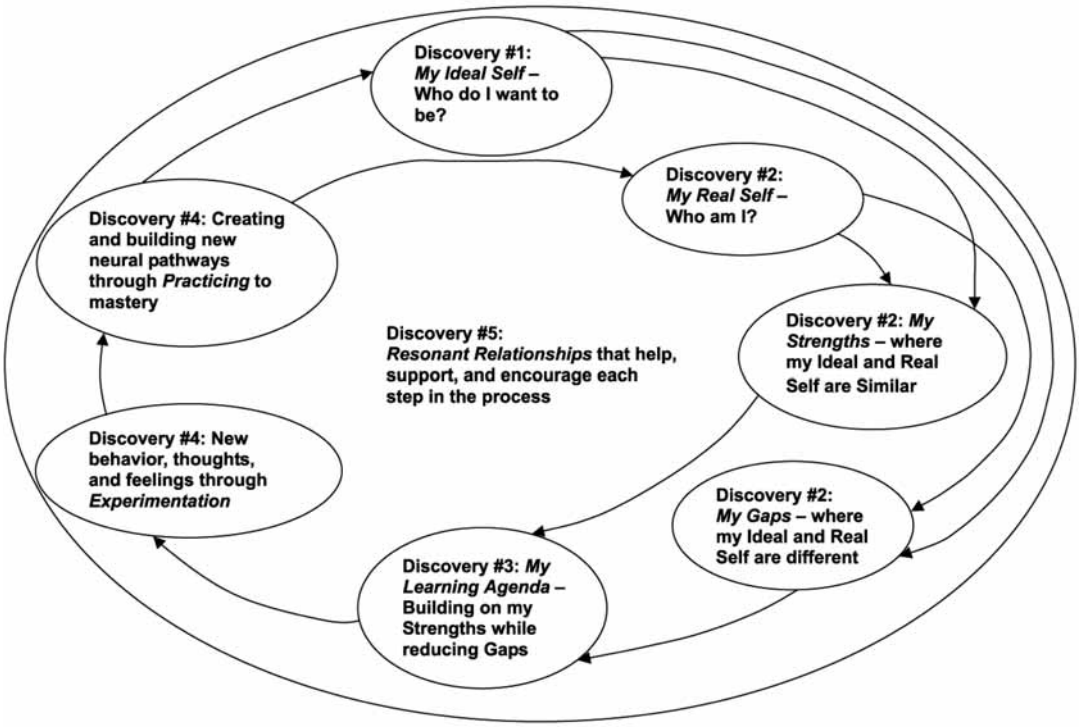

Fig. 2 Boyatzis' theory of self-directed learning [12]

the process of development and adaptation. As it is presented in the Fig. 2 [12], its discovery and development phases are:

- The first discovery: *My ideal I;* who do I want to be? Practicing new behaviors: Creating new neurological ways of improvement.
- The second discovery: *My real I;* who am I? What are my strengths and weaknesses? My strengths: Where do my ideal I and my real I match? My weaknesses: Where do my ideal I odd of my real I?
- The third discovery: *My learning plan;* how can I consolidate my strengths while reducing weaknesses? The learning plan is based on qualities and reduction of failures.
- The fourth discovery: *Experimenting and practicing* new behaviors, thoughts and feelings until they are perfect.
- The fifth discovery: *Developing relationships* that will give support and trust and provide the change.

3. Evaluating leadership development initiatives

It is always necessary to make an assessment that refers to a diagnosis of the current state of the individual or organization in terms of leadership capacity. These data could help one to learn clearly what needs to be improved or changed.

The most successful world's companies do not only give huge importance to development of leadership but put a great accent on trying to measure and quantify the effort invested in development of leadership skills. These companies "measure" the quantity of leader's gift needed in specific spheres, the degree of talent dissipation and make work catalogues, task catalogues and catalogues of leaders or managers who, by their nature have bigger influence on development of leadership skills. This gained knowledge is then used strategically, by rotating managers and moving them on certain positions providing them further training and improvement that will have positive effect on their development in a desired sphere.

"Evaluation helps people in organizations... make informed decisions about how to improve leadership development initiatives and examine the degree to which development goals have been accomplished and what work remains to be done" [13].

The data can be gathered from people through feedback, customer evaluations, etc. Also, leader can get information alone, as self-reflection, scores on personality inventories, etc. "Good assessment data also provides the motivation to close the gap between the current and an ideal future state. If no gap exists, this assessment data can still be helpful in terms of enhancing self-confidence" [11].

The main criteria for leadership development initiatives evaluating are:

- *Reactions,* that could be measured by post-experience questionnaires,
- *Learning,* that could be measured by tests or demonstration of skills,
- *Transfer,* that could be evaluated through self-reports, and ratings,
- *Results,* which could be measured through profitability, productivity, customer satisfaction, or staff morale.

Evaluation in the best practices includes stakeholders' involvement in all phases of process, consideration of multiple measures and perspectives, as well as clarifying of outcomes and well defined purpose of evaluation. Big mistakes in the evaluation can appear if the evaluation is not focused to outcomes, or does not pay attention on long-term impacts, or when evaluation is focused only to the factors that can be easily controlled. The worst use of evaluation is when the evaluation is used in the purpose to blame instead to learn and fix problems.

4. Conclusions

Leadership means competencies, knowledge, skills and expertise abilities. Although every manager needs to have some leader capabilities, real leaders belong, by their position, to the top management. They confront an increasingly complex milieu, with fast-moving world of changes. Large corporations, SMEs, low-cost competitors and entrepreneurs, they all share the same space. Every company needs to have good and effective managers that perform well at the operational or strategic level, as well as on functional or personnel issues. A company must help its managers to improve their capabilities, but the managers also must foster and improve their leadership skills themselves.

Although in available literature it is almost impossible to find examples of classification of human's skills, some classifications, like the model that exists for skills typologies, are possible. Then all skills could be classed into four groups, into intellectual, sensory, motor, and sensory-motor skills. In context of organizational behavior, intellectual skills are the most important. Now days, besides classical leadership training, attention is paid to emotional alignment with followers of the leaders, because only the absolutely committed employees become the key to success for the entire organization. All practical methods of development of leadership skills follow two main streams: the leadership training and the leadership development activities. The key feature of the majority of leadership development programs is that they are mainly focused on the skills, and less on knowledge and very rarely on abilities and personal qualities of holders of this business and social function. Leaders' self education and their self improvement are gaining in importance in today's doing business. In this sense, six steps in the process of self-building of leaders can be distinguished: mastering of the leadership competencies based on their own experience; getting rid of the myths and stereotypes about leadership; understanding of the treatment of leadership in their organization; self understanding; changing of the way of learning; and learning how to put up with failure.

The most important action in the leadership development is process of evaluation. It must cover: reactions, demonstration of skills, transfer and results measured through profitability above all. Also, evaluation has to be used on a proper way, always aiming to achieve better results in doing business.

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EXECUTIVE DASHBOARD SYSTEMS FOR EMERGENCY MANAGEMENT

The executive dashboard systems are a mainstream business intelligence technology. The main idea behind the executive dashboard systems is to deliver information about status of the organization by means of simple, abstract measures of performance and intuitive visualizations. One of the key concerns is preventing information overload. This is achieved by identifying key performance indicators – a set of measures that summarize the key aspects of the organization's performance. This paper investigates the application of the concept of the executive dashboard systems to the context of public emergency management. We identify the key differences between development of executive dashboard systems for profit-oriented organizations and for the multi-organizational network of the public emergency-management organizations. Based on these differences we deliver a set of implementation guidelines.

Keywords: *information systems, executive dashboard systems, emergency management, applications.*

1. Introduction

Dashboard systems are currently one of the most popular trends in business intelligence [1]. The main goal of an executive dashboard (ED) is to deliver real-time, relevant information on the current state of the organization for its executive-level decision makers. ED delivers only key, relevant information by means of intuitive visualizations on the computer screen, and – at the same time – prevents information overload through summarizing volumes of data into meaningful measures. Traditionally, it was the role of human assistants who interpreted volumes of data and prepared paper reports which included relevant and important information according to their judgment [2]. The main drawback of this process was a time delay – the report on executive's desk was at best one to two days old. The electronic executive dashboard designed for public emergency managers is intended to automate this rapid assessment process to produce more timely situation assessments of operational capacity and risk in emergency situations.

In this paper, we investigate the applicability of executive dashboard systems in a context of public emergency response. We address the basic question of whether and how executive dashboards can increase performance in decision making during crisis situations. Under crisis situations, the dynamics of the emergency response process is rapid and extremely complex, making the decision makers' task especially challenging. Some of the crucial problems that emergency managers face under those conditions relate to:

- lack of proper understanding of the broader context of the current situation,
- information overload that the decision makers experience at the time of crisis.

We discuss the challenges in applying ED to public emergency management. First, we provide a critical review of existing EDs, their strengths and weaknesses, and discuss the design process of such systems. Then, we discuss the suitability of EDs to the field of public emergency management. In particular, we identify basic differences between management of a single profit oriented organization and public, multi-organizational emergency response. We take the perspective of the development of socio-technical decision support system. Finally, we provide guidelines which should lead to a successful and efficient implementation of an ED and result in a measurable increase of effectiveness of emergency managers' decision process.

2. Background

The idea of digital dashboards evolved from the concept of decision support systems that emerged in the 1970s. Digital dashboards were particularly focused on data visualization for decision support. But the real beginning of the executive dashboard systems dates from around 1985 and is related to advances in computer displays and graphical user interfaces. Initially they were called executive information systems and were targeted only for the top

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executives of profit oriented organizations who, in fact, rarely used them. As a result, they had little significant impact. Advances in technology helped to make the development costs lower and widen access via the availability of a global computer network and advances in the World Wide Web. In the late 1990s, executive dashboards began appearing in updated forms. The same concept was replicated for lower level managers and this time proved to be more useful and accepted by potential users [3]. Although the current implementations of the executive dashboards used by the decision makers at multiple levels should be called managerial dashboards, the old name remains in use. A classic example of a successful executive dashboard is the system implemented for Southwest Airlines, or GE Aviation. This company developed proprietary software called "Digital Cockpit" to monitor the trends in aircraft spare parts business. In the late 1990s, Microsoft promoted a concept known as the Digital Nervous System and "digital dashboards" were included as one part of the system.

3. Technology

The EDs became possible when two technologies were introduced to organizations. First is the data warehouse – the unified, centralized, organization-wide database which is able to store and view the organization's data from different perspectives as well as to perform analysis on it. This has an important practical implication – it simplifies more advanced data analysis by the fact that all background technology for storing and retrieving information is already in place. The data are assumed to be already collected, stored and ready to use [4]. The ED development process can then focus on more important tasks – performing specialized analysis and presenting results to the user. The presence of the data warehouse has an important implication in terms of reduction of implementation costs – if the organization does not have a centralized data warehouse, the ED would require input data obtained from other sources and would most likely significantly raise the development cost and risks associated with it.

The second crucial technology for the ED is a web portal. The web portal is a concept closely related to the ED, which can be viewed as a specialized implementation of a web portal. A web portal is a web-based technology that provides a graphical interface to the organizational data warehouse and promotes sharing information within the organization. The web portals are intended for sharing data, promoting collaboration, and enhancing information exchange. Although a web portal is often restricted to the organizational intranet, often outsiders can have limited access to that portal – for example, customers and external suppliers. The EDs are in some sense highly specialized web portals intended for upper level executives. However, one element is different – an ED does not provide access to all data upfront – its key role is to transform large volumes of data into a small set of measures that summarize and interpret the data, rather than provide access to it. ED and portals use the World Wide Web to implement user interface. It's because the WWW lowers the cost of the implementation by using a unified, standard technology.

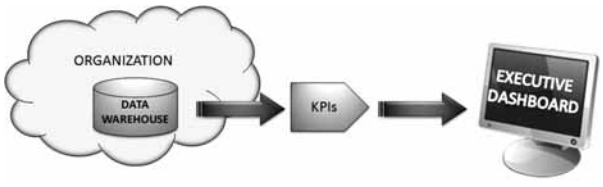

Fig. 1 The conceptual architecture of the executive dashboard

The architecture of an ED is relatively straightforward, and is shown in Fig. 1. A data warehouse serves as a source of real time information. The user interface implemented using WWW should be kept simple and provide information in a manner easy to comprehend. The heart, or rather brain, of the ED lies between the data warehouse and the user interface. Some authors refer to this element of the dashboard systems as a "filter".

The role of the "filter" is to query the data warehouse for relevant data and then transform it into key performance indicators (KPIs). The KPIs are the measures of the status of the organization that are derived from raw warehouse data and are displayed on the executive's dashboard screen. From a technical point of view, this architecture is relatively simple. Querying a database is a typical operation of a warehouse, and the web-based user interface is a standard technology, making implementation of both those elements relatively undemanding and inexpensive. The implementation of the filtering module is the most challenging part – in terms of technology and elicitation of the user needs, data requirements, and encoding the elicited knowledge about organization into quantitative models. However, the degree of complexity of the filter is left for the designer of the system. In the most naïve case, the KPIs can be taken directly from the data warehouse – for example, a trend in sales of a particular product over a specific period. On the other extreme, the filtering module can perform complex statistical analysis, data mining, machine learning, etc. Obviously, the former approach is appealing to the dashboard developers due to ease of implementation, however the risk is that it can result with KPIs that do not capture the user's needs adequately.

4. Goals of the Executive Dashboard System

An ED is intended to support the decision making process by providing summarized data and guiding their focus to areas that need attention. Alex Kirtland [5] identifies three tasks that an ED system should:

- answer fundamental questions about the business or business unit,
- alert the user to issues or problems in such areas as production, sales and revenue,
- help make decisions that impact the business.

The second goal requires more explanation because not all EDs designs include explicit warning system.

The implementation requires more effort than merely querying the data warehouse. The most naïve implementation of an ED

can be achieved by identifying several key performance indicators that are already in the data warehouse. Including a warning system in the ED requires knowledge elicitation from the domain experts and the users of the system, which significantly raises the implementation complexity and costs. That would explain why many developed applications do not have the warning system included.

The third goal is to provide sufficient information for making decisions. The decisions should not be made on the basis of generalized indicators, rather based on thorough analysis. Although the primary goal of the ED is to deliver only limited and relevant information, a properly developed ED should serve as an entry point for obtaining more detailed data and/or information that is really needed for making informed decisions.

5. Key performance indicators

The main challenge with developing EDs can be summarized by the following question: *what information should be presented for the decision makers on the executive dashboard?* In no measure does this question have a simple or obvious answer. On the contrary, this problem is difficult to address because the answer always depends on the particular context in which the dashboard is intended to be used – every dashboard should be unique and customized for a particular user's needs. Therefore, EDs are referred in the business literature as a solution rather than a product. Every time the executive dashboard is implemented, the most critical issue is the decision about what data and information should be used to define KPIs.

An ED should incorporate multiple KPIs that measure different aspects of the organization's condition. They are strictly dependent on the organization's resources, goals, available data, etc. as well as on a particular decision maker's responsibilities and data needs. Examples of KPIs can be: number of employees, revenue for particular product, number of available beds in a hospital, supplies in the warehouse, etc. The very difficulty of developing EDs is in identifying key performance indicators – to do so properly, one should have a deep understanding of a social system in organization, its goals, responsibilities division, etc. and the available data in the organization's data warehouse that is relevant. The selection of the KPIs determines the practical success of the executive dashboard and cannot be underestimated. Additionally, the definitions of KPIs are not necessarily static – for example, when the goals of the organization change, the KPIs should be adjusted accordingly.

6. Visualization

Although the user interface for the dashboard systems is not a primary concern of this paper, there are aspects that require discussion. A graphical user interface used for the dashboard should be implemented using WWW technology – to enable users accessing the dashboard from a web browser from any location.

In terms of design, the executive dashboard does not differ much from any other typical user interface and all good design practices for the web interfaces hold true. The key performance indicators should be presented in a highly visual but easy to comprehend way – preferably with graphs, charts and similar visual images. The most important criteria should be clarity and readability. For example, practitioners indicate that for example 3-D graphs are not suitable for the executive dashboards because in practice they are more difficult to read and are harder to comprehend than traditional 2-D graphs.

The convenient ability to drill down should be taken into account when designing the graphical user interface. One of the most valuable features of EDs is the ability to reach more detailed information when needed. In that case, the dashboard interface would serve as a starting point to the organizational portal, or other sources of data. In general, the graphical design should include links and database querying abilities. Figs. 2 to 4 show some examples of executive dashboard interfaces.

The EDs originated as an application of information technology (IT) for decision support in the context of business organiza-

Fig. 2 Interface example of Oracle/PeopleSoft Enterprise Service Dashboard

Fig. 3 Example of business dashboard interface MicroStrategy 8.1

Fig. 4 Example of dashboard interface of PacketTrap pt360

tions. They were created with the business top executives in mind. There were two technology developments in business organizations that become enablers for affordable development of the ED systems: centralized data warehouses populated with large volumes of real-time data, and the development of the intranets based on WWW technology. The practical value of the dashboard systems became evident when they were made available for the greater number of the decision makers in the organization, not only top executives.

7. Dashboard System for Emergency Management

Application of the ED concept to the context of the public emergency management entails specific adjustments. The question is whether the EDs, which are main stream in business organizations, have the potential to be successfully adopted in the public sector. It is well recognized that profit oriented organizations are more advanced than public organizations in absorbing new technologies. This is especially true for technologies that increase organizational efficiency in terms of information flow and decision making. Although this trend seems unfavorable for the public sector, there are potential benefits for the public organizations – such as learning from somebody else's mistakes. We summarized the lessons learned during implementation of executive dashboards in business organizations. In the process of transferring those lessons to public organizations, we recognize that some significant differences between profit and public organizations exist. Those differences can have direct implications on the implementation and acceptance of the EDs in the public sector. Our particular interest is the application of EDs, as an instrument to enhance decision-making for public emergency managers in rapidly changing environments. Due to different goals within emergency management and emergency operations, we identify a new set of tasks that an ED for emergency management should support:

- deliver situational awareness in a complex, multi-organizational environment,
- alert the user to emerging issues and problems in rapidly changing crisis environment,
- develop valid estimates of the changing status of the operational environment in complex highly interdependent systems,
- deliver timely information to improve decision making.

8. Emergency Management as Multi-organizational Effort

One of the key differences between dashboard systems discussed earlier and public emergency management is that emergency management requires a 'common operating picture' for effective coordination among many organizations, while a business operates primarily as a single organization. This fact has extremely important implications.

The term "multiple organizations" implies, in practice, an environment with multiple data sources that are not connected to each other. Implementing a dashboard system for emergency management requires accessing information from multiple sources. This process creates additional technical challenges (typically multiple protocols, security restrictions) and increases the implementation cost. But the technical aspect is only a part the problem. Even more challenging are the legal, cultural and organizational aspects. What information can be made available to the public and which organizations have access to what information? These questions should be determined in the design phase that involves executives and lawyers of interested parties. Nevertheless, one should expect additional challenges with transferring data between organizations – additional security measures will be required rising implementation costs. However, the very nature of the KPIs and their most likely abstract meaning can be actually a great help. For example, hospitals are extremely cautious about their patient data, but providing only the number of beds available may not necessarily raise strong objections.

The other aspect relates to organizational issues such as getting all the essential parties to work collectively on creating the ED. Unlike a business setting, neither there is one authority that can make decisions nor do all the organizations involved share the same goals. One example of potential distortion is pushing the design of the system to meet local organizational goals rather than common goals, or simply lack cooperation.

The number of organizations involved in emergency management has an important implication in terms of identifying KPIs. The task of identifying KPIs requires gathering requirements from all involved organizations. Usually, most of the KPIs for given organization will be "local" in that they will be only KPIs for the same organization that owns the data source. This will imply that the work is split between obtaining global KPIs and local KPIs. But the process of defining KPIs for a multi-organizational system can provide very valuable insights into the nature of the existing emergency response system and can lead to significant improvements of the system.

Developing an ED for public emergency management requires focusing on the aspects that distinguish its implementation from an ED for business organizations.

9. Design

The first step in the implementation of an ED for public emergency management should be the identification of organizations involved in emergency operations. In comparison to a single profit-oriented organization, the organizations involved in the development of the ED system can be of two types:

- customers – organizations that actually are users of the system, and
- data sources – organizations that provide information to the system.

Most likely, organizations that are customers will also be providers of the primary data sources. Because complex emergency response operations involve multiple organizations that create nontrivial interdependencies, information owned by the customer organizations will be insufficient for developing a common operational picture. External data from non-customer organizations is also needed. Sharing data by organizations that are not customers poses practical challenges – for example utility companies that are legally obliged to share their data during emergency situations may not be willing to cooperate, or at least may slow the development because of lack of direct benefits for them.

Having identified organizations that are customers, the next step is to define who the users of the system within these organizations are. The lessons from the early development of EDs systems are clear – the users should be the decision makers that most likely will use the system – not necessarily the top managers. In the initial phase of the implementation (assuming the project is developed as a scalable solution) the first batch of users should be those who express interest in, or need for, using the ED in their daily roles. Unlike many other socio-technical information systems, an ED provides a solution targeted for an individual user or agency, and the idea is to increase efficiency of organization by providing support for individual decision makers who are influential within the organization. Therefore, the ED should be highly individualized and easily customizable.

One of the challenges that developers face is the availability of required information. Unlike the case of a single organization, the organizations involved in public emergency management do not have a single data warehouse, and most likely do not have protocols for sharing digital information. The fact that public organizations are significantly behind profit oriented organizations in terms of use of IT is a disadvantage. In developing EDs for profit-oriented organizations, the presence of an organization-wide data warehouse is assumed as a fact. In public emergency management, such an assumption would be unrealistic. Developing an ED for a set of public organizations implies the development of the data information sharing system among these organizations. This task needs to be included within the framework of the ED development, adding to the complexity, costs and challenges. One practical benefit of this process is that the decision makers should recognize the importance of multi-organizational information sharing, leading to understanding the need for development of a comprehensive data sharing system and common database among organizations involved in emergency management.

Before the common databases for public organizations become a reality, the EDs need to implement their own means of accessing necessary information from various organizations and data sources. Obviously, this increases the cost of development and/or inevitably leads to difficult compromises. The discrepancy will arise between the information that is needed and the information that is available. One of the dangers in this process is that the definition of KPIs will be driven by the available information, instead of the critical information identified by the decision makers.

A basic requirement for any information technology system for emergency management is for the system to be used on daily basis – only then does the system stand a chance of being used during a crisis situation. A system that is not used on everyday basis, but is intended to be used during a crisis, will not serve its purpose – lack of familiarity and proficiency with the system will make users abandon it when the real crisis happens, because they will treat it as an additional burden. Encouraging the managers to use the ED on a daily basis should be a primary objective. One technique that can be borrowed from the experience of profit organizations is the idea of combining within the ED other daily functionality such as email, organization news board, learning modules, etc.

In terms of visualization, there is little difference between profit organizations and emergency management. One factor that the developers should take into account is that emergency managers are working under extreme time pressure and do not have much time to spend in front of a computer screen. Another lesson learned from implementing decision support systems for emergency management is the importance of producing hardcopies of reports. Although we like to think that we are in the digital era, many decision makers find reading from paper more convenient than from the computer screen. Especially in crisis situations, it may not be only about the users' preferences – in these situations access to computers by some users can be limited and paper copies are simply a necessity.

A typical practicing emergency manager is a very busy person, especially in crisis time. To increase usability of the ED, it is of a great practical value to gain an understanding of how emergency managers access and manage information. Learning how decision makers are currently obtaining necessary information can lead to addressing their information needs in better ways. An ED should fit into these patterns, and minimize the burden on the users caused by learning a new tool and changing their practices whenever it is not necessary.

Crisis management is a very complex and dynamic process. The challenge that socio-technical systems face is ensuring that the data are up to date and credible. The data quality in the data base affects directly KPIs presented by the ED. In the context of a rapidly changing situation, it would be necessary to present not only KPI values, but also annotate them with some form of time

stamps. This would provide at least some hints for the decision maker on how current and reliable the KPIs are.

Another valuable feature of an ED is the ability to drill down. The ED presents the information about the condition of the organization on a single screen. This is desired from the perspective of providing high-level, easy to comprehend information on broader context, but, it may be too abstract to make informed decisions. A solution is to provide some mechanism to drill down on selected aspects of the data – to access more detailed information. For example, by clicking on the red hospital icon, the user can access the data on this particular hospital and learn the details of the problem. The drill down feature can be relatively hard to implement when the data sources belong to various organizations. The alternative solution, that can serve a similar purpose, may be to provide contact information for other organizations. That is, clicking on the icon would provide pop-up window with contact information to a representative who can provide required information.

10. Costs

The question of costs associated with the implementation of an IT system is central, followed by whether the ratio between costs and benefits justify the investment. We do not address the question whether the implementation of an ED is justified; this question can be answered with knowledge of scope and details of the planned project. However, it is possible to discuss how the costs are distributed and what parts of the project can be especially costly.

In general, the ED is a relatively low cost enterprise, which greatly benefits from already existing infrastructure (such as a data warehouse or organization's portal). This may be true for the profit-oriented organizations, but costs rise when the required infrastructure is not in place and would be required to implement it along with the ED. A similar problem arises, when communication and data exchange protocols between multiple organizations are not available, but they are required for the ED, for effectiveness and evaluation of these processes uses, see [6] and [7].

Implementation expenses are not the only cost associated with the dashboard system [8]. Maintenance costs are also high, and according to developers of ED systems for profit-oriented organizations, the yearly maintenance costs are approximately equal to the implementation costs. This is caused by this fact, that the KPIs are in most cases not static and require continuous adjustments. Additionally, the ED is in practice highly dependent on the other IT infrastructure within the organization, therefore changes in one system can influence some of KPIs delivered to the ED. Organizations constantly improve their IT infrastructure and in many cases this effort opens opportunities for improving the dashboard system. If the ED includes an alert/warning system, such a system requires frequent calibration that should be performed with active participation of the user.

11. Intelligent Dashboards

EDs differ with the complexity of analysis that leads to KPIs. In the most naïve case, the dashboard presents simply pre-selected information obtained directly from the database, and leaves the task of data analysis completely on the decision makers. More complex EDs allow the user to set thresholds on each individual KPI that, once reached, trigger a warning message, or inform the decision maker in some other form. The question arises whether one could provide KPIs that are the result of more advanced analysis (based on for example on intelligent systems, knowledge engineering, or statistical models) to the decision maker. Such an automated system can prove to be valuable [9], especially when the decision maker is making decisions in a rapidly changing environment and under time pressure, as it is a case in the emergency management.

Successful applications of artificial intelligence, data mining, or machine learning share one common property – they solve problems for relatively small, specific domains, and not for general, complex domains. The EDs operate on very complex domains therefore the expectations should not be high. Our consideration is what data analysis techniques can be used, and which of them are most promising. We assume that input is the set of KPIs of our ED. The ED, by definition is a system that is used to make generalizations on performance of the organization therefore it is operating in a highly complex system. This is the reason why it is a challenging process.

Emergency management is a complex, dynamic process, and therefore, replacing human decision making by any form of automated process is far beyond the reach of current technology. By definition, an emergency is a sudden, unusual and unique situation. For example, machine learning techniques require large volumes of data that relate to repeated, similar situations. Collecting such data is virtually impossible for emergency situations – because every emergency is unique and far more complex to close it in an input-output description. However, the machine learning methods have been extremely successful in applications to anomaly detection – and can be useful for KPIs that are intended to provide early warning by detecting some anomalies in everyday data streams. Once the anomaly is detected, the problem is with the interpretation of the consequent data – and machine learning methods are likely to fail. The models that are built from the combination of historical data and knowledge elicitation form human experts are more promising in terms of defining models for KPIs. In particular, Bayesian networks are suitable for such fusion of data and knowledge. One should appreciate that the actions taken by a decision maker are in most cases complex processes not easily defined by quantitative models – for example evacuation of the elderly population using local school buses to a stadium located several miles away is not at all easy to capture in a quantitative form. The automated approach is not an appropriate tool for analyzing KPIs. However, automated methods may be valuable for solving some sub-problems in the domain or interpreting a specific subset of KPIs.

The KPIs are defined by a domain expert, and in this sense the ED is an expert system, as an expert knowledge is already placed in the system. In the same way, use of experts' knowledge can be a promising alternative to data-driven approaches. For example, a model based approach, where the model is built by an expert, seems a natural extension of the ED. In some sense, assuming thresholds on KPIs and deploying warnings can be viewed as a kind of model built by an expert. For more work see [10] and [11].

Likely the best that we can do is to provide an intelligent decision support system that can assist a decision maker by providing suggestions and highlighting elements that, according to the system, are relevant and should receive user's attention [12]. At best, some sub-tasks in this process can be automated the system can aim to detect user mistakes, rather than solving the problem [13].

12. Conclusion

The EDs were originally designed to deliver key information about the condition of the organization for the key organization's decision makers, and they were developed for profit-oriented organizations. The main goal is to deliver key information in a timely manner for a busy individual and make the presentation of this information as clear as possible. The concept seems to be well suited for emergency managers. An ED is a tool for leaders – they provide information for the key decision makers, and in that sense each ED is unique, designed to deliver information relevant to the decision maker's role.

From a technical point of view, EDs extensively utilize databases available within the organization, and in some sense their success depends on how well an organization's information technology infrastructure is developed. In particular, the central data warehouse is the key technology, which when present, can immensely lower cost of the implementation of the ED. The ED concept is an extension of the portals – the web based information system serving the needs of information exchange within the organization. Although the ED uses the same technology as a portal, the goals, users and scope are significantly different.

In terms of implementation, the EDs are challenging for several reasons. One of them is the audience for which they are intended – the upper level managers are not easily accessible, and understanding their individual and often unique objectives is not an easy task. Moreover, those objectives change over time.

All practicing developers of EDs agree that the most important element of the implementation is identification of the key performance indicators during the design phase. If the KPIs are selected inappropriately, in the sense that they do not reflect real needs of the decision maker, the ED is virtually guaranteed to be ignored by the user. Special attention should be paid to the process of identifying the KPIs for the system. The same problem can be viewed from a different perspective: the real needs of the decision makers should drive selections of KPIs, not any other factor such as easy availability of some data. This is especially important in case of the EDs, because their users are highly intolerant to irrelevant information.

The EDs started as a tool developed in profit oriented organizations and they are well suited in this setting. When trying to apply this concept to emergency management, one should be aware of the following crucial differences:

- Non-profit emergency management is a multi-organizational task, while a typical setting for which the EDs were developed is a single profit organization.
- Multi-organizational setting raises the costs of the implementation, because of the lack of unified IT infrastructure and additional cost of implementing secure channels of communication.
- Many organizations, each with different goals, can result in political influences on the project.
- In general, public organizations are significantly behind in terms of the IT adaptation for their needs.

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Martin Hrinko – Svetla Fiserova *

STUDY ON LIGHTING PARAMETERS OF AUDIBLE AND VISUAL WARNING DEVICES USED BY THE POLICE OF THE CZECH REPUBLIC

In the article, the authors are concerned with the lighting parameters of audible and visual warning devices used by the Police of the Czech Republic in reaction to European legislation on this issue. The aim of this article is to explain the principle itself and the properties of radiation of light with its photometric specifications that can be evaluated and perceived by human eye, including subsequent negative effects on human health. The article presents a study of evaluation of measured values of luminance, luminous intensity and other photometric quantities of selected audible and visual warning devices used by the Police of the Czech Republic.

Keywords: *Police of the Czech Republic, photometric quantities, lightbars, beacons, audible and visual warning device*

1. Introduction

Recently, special audible and visual warning devices (beacons, lightbars) of the Police of the Czech Republic, and also of the other emergency services all over the world have been discussed widely, above all in relation to the colour of light emitted and the continuous development in the area of used luminaires and materials of which these luminaires are made with the aim to use electricity more effectively, to improve visibility in terrain and to achieve longer life, similarly to the other branches of industry. These are the main reasons why the authors of this article are concerned with lighting parameters of police audible and visual warning devices of types used in many cases by the other safety and security services.

The problems of audible and visual warning devices used in the Czech Republic and in the world and the comparison of them have not been dealt with comprehensively yet; greater attention should be paid to these questions owing to their importance for the informing of citizens in critical situations, e.g. when human lives have to be saved. From the point of view of assessment of lighting parameters, the knowledge of the general physical principle of propagation of visible light and of photometric quantities, which are of importance to the comparison of photometric characteristics of several types of audible and visual warning devices with each other, is significant. An integral part of assessment is also the knowledge of the area of vision physiology and of negative health effects of inadequate exposure of human organism to the radiation of visible light. The assessment also has to respect the requirements of relevant valid legislation and international standards. [1–12] Moreover, the proper verification and the evaluation of photometric quantities have to be in accordance with the requirements of relevant international standards. All these aspects were taken into account when dealing with the joint project on the basis of cooperation between the Faculty of Safety Engineering, VSB – Technical University of Ostrava and the Regional Headquarters of Police of the Moravian-Silesian Region of Police of the Czech Republic (henceforth referred to as PCR). The project was simultaneously directed towards the acoustic parameters of audible and visual warning devices of PCR and was implemented in the years 2009 and 2010. Its results form the content of two Master's theses thematically focused like that; the authors being students in the field of Safety Engineering of Faculty of Safety Engineering of VŠB – Technical University of Ostrava.

2. Properties of Visible Light in Relation to Human Perception of Audible and Visual Warning Devices

Light composed only of waves having one wavelength or little different narrow-band wavelengths is perceived by human eye as coloured light. Visible radiation is part of a spectrum of non-ionizing radiation in the range from 3.9×1014 Hz to 7.9×1014 Hz.

The spectrum is a physical card of colours produced by the decomposition of white light. The colour stimulus is radiation characterised by a certain spectral composition and luminous flux or luminance, which penetrates into the observer's eye and induces a colour sensation. The same visual perception which would be created by a single colour of the spectrum can be obtained by mixing two or more other spectral colours that do not contain the wavelength of the resultant colour at all.

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Colour properties of light are called chromaticity; it is given by the spectral composition of radiation emitted by a source and is usually given in trichromatic coordinates. A typical example is the observation of coloured signal lights that are often perceived as point sources. In the trichromatic system, the fact that by additive mixing three measured colour stimuli a perception of any colour can be obtained is used as a basis [13].

Luminous intensity **I [cd]** is the fundamental photometric quantity expressing the property of a light source (point). It describes the spatial distribution of the luminous flux of a source or a luminaire. For a point source, the luminous intensity is defined as a ratio of the luminous flux emitted by the source in an elementary solid angle containing the given direction to the size of this solid angle. As a point source almost every luminaire the largest dimension of which a is negligible in relation to the distance l between the source and the check point, $l \geq 10\ a$, can be regarded. Luminous intensity distribution curves are sections through the surface of luminous intensity distribution that run through the light centre of the luminaire. The surface of luminous intensity distribution is the surface determined by the endpoints of all luminous intensity vectors that start at the light centre of the luminaire. In the case of point sources of light, the distribution of luminous flux can also be obtained from luminous intensity distribution curves. In catalogue cards, luminous intensity distribution curves are plotted in polar or rectangular coordinates.

Luminance **L [cd.m^{-2}]** is determined as the luminous flux that is emitted from, impinges on or passes through the elementary surface dS in relation to the projection of the elementary surface dS onto a plane perpendicular to the direction given by the axis of elementary solid angle d.

Radiation in the visible part of the spectrum is absolutely necessary for life, but the high intensity of light may damage the retina. The effects can be, depending on the size of exposure, desirable for as well as harmful to the organism. By radiation incident on the retina two types of reactions occur, namely photothermal and photochemical. The photothermal reaction takes place after heating the retina by 10 °C; photochemical damage appears after the incidence of intensive blue-light radiation on a photoreceptor that will die. A similar phenomenon may occur after illumination of the eye with a white LED, which is mostly composed of intensive blue and luminophor. A possibility of eye damage is however lower, because luminophor emits light in the range of maximal sensitivity of the eye, and after illumination, the eye adapts quickly by changing the diameter of aperture of the iris leading to a 16-fold decrease in the area, which will considerably reduce the amount of light on the retina. To the blue colour the eye is minimally sensitive, and thus the iris reacts minimally. Like other forms of energy, the non-ionizing radiation is also able to react with biological systems. The effect depends on many physical parameters, especially on the wavelength of radiation that determines the place where the radiation can be absorbed, and on the intensity of radiation that is crucial to the degree of radiation effect [13].

3. Audible and Visual Warning Devices from the Point of View of Czech Legislation

The most significant law regulating the use of audible and visual warning devices in the CR is Act No. 361/2000 Coll., on road traffic (Road Traffic Act). With a special audible warning device supplemented with a special blue warning light, the following vehicles can be equipped [3]:

- those of Ministry of the Interior used by the police and marked according to a special legal regulation
- of Prison Service
- of Military Police marked according to a special legal regulation
- of Municipal Police that are determined by the municipality concerned
- of Fire Rescue Service
- of Mine Rescue Service
- of gas breakdown services
- emergency medical services and transport of sick and injured persons and that of expectant mothers
- of armed forces used with military rescue forces for the performance of humanitarian tasks of civil protection
- of customs authority marked according to a special legal regulation.

It is necessary to emphasise that in the CR *merely blue beacons without a possibility of combining with other colours* are allowed in all vehicles of so-called rescue and security services.

The homologation of audible and visual warning devices used by the Police of the CR is carried out at the Electrotechnical Testing Institute in accordance with the requirements of the binding European EEC Directive No. 65.

The other principles for the use of audible and visual warning devices by Czech policemen follow from a binding instruction of the Police President from the year 2002. The instruction determined uniform principles for the use of audible and visual warning devices. In the instruction of the Police President, lightbars and beacons are regarded as special audible and visual warning devices. The use of a special audible and visual warning device is understood as giving a blue light signal, which can be supplemented with a special audible warning signal. Both can be used in combination with a public address system. The lightbar/beacon for this purpose is a detachable or attached device mounted on a vehicle of the Police of the CR. The lightbar/beacon can be used merely in the course of fulfilling tasks associated with the performance of special duties; police vehicles having the right of way.

The majority of vehicles of the Police of the CR use audible and visual warning devices supplied by the company Zdenek Holomy Electronics (henceforth referred to as company). The assortment of products offered by the company on the market for various kinds of sources of light is wide. For this reason, for every source of light the authors selected one representative of the devices that use this source. The first device is designated VNK 530 (incandescent lamp set) – Fig. 1. The type of the second is VSK 012F (discharge lamp set) – Fig. 2. Another flashing lightbar is desig-

Fig. 1 Lightbar VNK 530

Fig. 2 Beacon VSK 012F

Fig. 3 VSL 012FL

Fig. 4 VSL 012LV

nated VSL 012FL (linear discharge lamp set) – Fig. 3, and a lightbar with LEDs – diode lightbar VSL 012LV – Fig. 4. In individual types, a large number of configurations of the light source itself, an acoustic device and additional word markings, such as POLICIE, STOP, and others exist. For this reason, the authors of the article always supplemented the selected device with a figure showing a corresponding measurement configuration. The presented devices are compatible as far as the control device, e.g. AZJ 530 A CO, is concerned [14].

4. Assessment of Lighting Parameters of Audible and Visual Warning Devices

In the case of all four above-mentioned sets, screening measurements of selected photometric quantities were made using professional measuring devices in the framework of the implemented study, namely *luminance, illuminance (instantaneous luminance) and flash frequency*. Luminance measurement was carried out beyond the requirements of EEC Directive No. 65. Additionally, measurements of trichromatic components and colour coordinates were taken merely with the selected lightbars and beacon.

The measurements were made both in real conditions of operation, i.e. in the shadow of a bridge structure in a D1 motorway segment closed to the traffic, in a darkened room and in a light laboratory in the HARD building of Faculty of Electrical Engineering and Computer Science of VSB – Technical University of Ostrava.

For the screening measurements, standard requirements of relevant international standards were satisfied; for technical reasons, it was not possible to take all measurements as prescribed by the EEC Directive No. 65.

With reference to the fact that the sets are different in the used light source and design, the comparison of these sets with each other is difficult. Photometric specifications and other technical parameters, such as set power input, etc. can be quantitatively compared. The compared photometric specifications are based on the requirements of EEC Directive No. 65. In the case of approved types, they are above all luminous intensity, colour coordinates and flash frequency. Beyond the Directive, luminance measurements were made with all types of the given audible and visual warning devices under the same conditions. All measurements should be carried out in dark areas without a possibility of influencing by stray light. The measurements were always done on one optical unit (i.e. one beacon in the lightbar) owing to the symmetry of the set. There are certain deviations in photometric properties between the same types manufactured in series as permitted by the EEC Directive No. 65.

For the measurement of trichromatic components a chromameter CS-100A from the company Minolta was selected. The chromameter is a professional measuring device for measuring the integral colour characteristics of light; likewise a luminance meter, it measures the properties of light in a delimited solid angle - aperture of the objective. In addition to the luminance in cd×m-2, it can measure the colour coordinates x, y. Besides the absolute measurements also different values, i.e. those related to the value stored in the memory, can be measured. The device communicates with a PC by means of a serial interface and is able to print directly the measured data on a special printer. The standard measurement angle is 1°, which corresponds to the solid angle of about 0.00024 sr. The uncertainty of the measurement of colour coordinates (x, y): ± 0.004. In the course of last measurement, the colour coordinates were measured using a filament spectrometer AvaSpect 2048-USB1. For the measurement of luminance, a luminance meter LS 100 from the company Minolta was selected. The luminous intensity was measured using a universal measuring device Almemo 2590-4S from the manufacturer Ahlborn, with which a photocell FL A613-VL (sensor of illuminance with a range of 26 000 lx – the manufacturer specifies the accuracy as 5% of measured value) was used. The flash frequency was determined by visual observation and recording of the number of flashes within a certain time or by recording the behaviour of luminous intensity with a luxmeter, where the development of flash peaks of beacons with time can be seen. For the measurement of the on time of the flashes, a universal measuring device Almemo 2590-4S was used [13].

The measurement uncertainties can come from the comparison of photometric properties of the same special warning devices, their deviations from the approved type in the case of manufacturing in series; the uncertainty is further given by the precision of measuring devices, by human errors (location of the device during measurements, misread values, unsuitably selected measuring range), crucial changes in environmental conditions (temperature, pressure, humidity, undesirable substances in environments, and others), interference and miscalibration. A change in supply voltage of the set and the on time of the flash, which influences the response of measuring and evaluation unit of the device, will play a great role. Measurement results will also be influenced by the reflectance of the background in the room or space where measurements will be taken. As for the measuring devices, the influence of a systematic error, i.e. uncertainty of the type B, is considered.

5. Measurement Results and Discussion

The measurements were carried out repeatedly under all the above-mentioned conditions of measurement from the 16th April 2009 to the 30th March 2010. The luminous intensity was recalculated according to the square law: $I = E \cdot r^2$. Values of luminous intensity and luminance are rounded to integers to obtain clear results. The luminance of areas of equal size measured at any distance from the areas should be the same. A difference in luminance measured at a distance of 1,014 mm and that of 2,000 mm is caused by measuring a larger part of luminaire area where not only a light active part of the luminaire might be measured. From a one minute record, the maximum value of instantaneous luminous intensity of every set was always selected. In the case of luminance of the sets, the greatest value of ten measured values was selected. Maximum measured values of selected photometric quantities are compared in Table 1 which summarises the results of all taken measurements under the above-presented conditions on four various types of audible and visual warning devices.

Maximum obtained values of photometric quantities Table 1

Type of set	Maximum instantaneous luminous intensity	Measured flash frequency	Maximum value of luminance measured at 1 014 mm distance	Maximum value of luminance measured at 2 000 mm distance
	[cd]	[Hz]	[cd/m^2]	[cd/m^2]
VNK 530	6 774 ± 389	0.9	18 120 ± 364	16 740 ± 337
VSK 012F	-	2.5	11 680 ± 236	10 330 ± 209
VSL 012FL	4 184 ± 209	2	12 460 ± 251	11 340 ± 229
VSL 012LV	261 ± 13	2.85	> 49 990	33 210 ± 666

The observed values of photometric quantities of various types of audible and visual warning devices used on vehicles of the Police of the Czech Republic and experience achieved when dealing with the project are a basis for the following summary.

The required parameters of effective luminous intensity can be obtained either by extending the on time of the flash to the maximum possible admissible value, or by increasing the instantaneous luminous intensity at a relatively short on time of the flash. During outdoor measurement, the highest instantaneous luminous intensity was achieved by the lightbar VNK 530 – incandescent lamp set, namely 6 774 ± 339 cd. At the distance of 1 014 mm, the luminance was 18 120 ± 364 cd/m^2. Measurements in outdoor spaces were considerably affected by environmental conditions, especially by daylight radiation and other sources of light and reflections typical

of real conditions. More accurate results would be achieved by the complete elimination of stray light and reflections. In the course of measurements in the darkened room, the highest luminance was achieved by VSL 012LV diode lightbars with which the measuring range of luminance meter was exceeded. The advantage of sets using the LED technologies is the colour of emitted light that does not depend on the passage through a filter leading to the colouring of emitted light. With regard to the high values of luminance that may be in the case of exposed persons a source of glare, eye protection should be considered. For eye protection during measurements and repairs it would be suitable to use absorption colour filters and/or shields in the case of discharge lamp sets and above all in the case of efficient sets utilizing LED technologies. Attainable luminous intensities are so high that there is a need to observe the limit of maximum values prescribed by the EEC Directive No. 65, because they may be exceeded when the main "beacon" is used with additional lights, which results in the glare of drivers mainly at night [13].

Measurements done in the field, in the darkened room and the light laboratory using stationary measuring devices are, in spite of their relatively large extent, the basis for the further verification of photometric properties of audible and visual warning devices used by the Police of the CR and for the comparison of lighting parameters of sets utilizing either a combination of blue and red colour or only the red colour. By combining the flashes of the device emitting blue and red light, effective visibility in fog and sunny weather could be achieved.

6. Conclusion

The European EEC Directive No. 65 allows the use of the red colour of light. The red colour of light is generally used to indicate a ban, emergency, stop, failure or any danger, and from the point of view of meaning of colours of visual signals, is superior to the blue colour of light. The blue colour draws extraordinary attention, indicates the right of way and actions leading to solving some difficult situation. The colour of the light emitted by police "lightbars/beacons" is prescribed merely by national legislation. For instance, in Slovakia vehicles having the right of way can use special warning lights of red or blue colour or of their combination. In the CR the public is accustomed to the use of this colour of light emitted by lightbars. The selection of the colour of light of lightbars is a very sensitive issue. If merely the red colour of police lightbars was used, the observer could confuse these devices with the brake lights of cars. However, by a suitable combination of flashes of the device emitting blue and red colours of light, effective visibility in fog and sunny weather could be achieved. Moreover, the red colour of light or the blue colour of light could be used depending on the circumstances of use of the audible and visual warning devices according to the severity of situation being solved.

The issue of photometric properties of visual and audible warning devices is really topical in connection with the continuous development of new technologies in the area of used materials, luminaires and energy sources as well.

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