VoIP LAN/MAN traffic analysis for NGN QoS management

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Strict requirement is emphasized regarding QoS guarantees of the NGN (Next Generation Network) networks today. DiffServ mechanism is applied mostly for classification of protocol data units of real time and conventional information streams in LAN/MAN environment. The dependence of VoIP traffic characteristics of the delay and the jitter sensitive IP telephony vs. voice codec applied can be considered an exciting scientific question. We analyze Ethernet traffic generated by G.711, G.723, G.728 and Wideband (G.722) voice codecs. The self similar, fractal and multifractal properties of popular TCP based services (http, ftp, telnet, etc.) in LAN/MAN environment are well known for several decades. In this paper, we study the effect of UDP based current voice mechanisms on self similarity of the Ethernet data traffic. UDP traffics of the IP phones are evaluated in congested and congestionless environment using sophisticated methods of entropy and wavelet analysis. A new and efficient evaluation method, named ON/(ON+OFF) transformation is applied to the characterization of VoIP traffic.

1. Introduction

One of the most important time critical services of current and also of ITU-T NGN (Next Generation Network) communication networks in the near future is voice traffic. The VoIP (Voice over IP) technology radically transformed also the cost of telephone service and the behavior of subscribers. IP phone service exploits efficiently the Internet based network infrastructure and approximates the quality of PSTN traffic services. The best-effort transmission mechanism of IP networks cannot assure guarantees for delay sensitive voice traffic, hence for successful utilization of VoIP QoS techniques are needed among the end nodes. The evaluation characteristics of modeling aggregated voice and other type of traffics implies selection of optimal QoS mechanisms. The network traffic coming from a voice source depends strongly on the utilized voice codec type. These codecs are grouped into two classes: constant bit rate mechanisms (e.g. G.711), as well as silence supression mechanisms based on repeating ON and OFF periods of activity (e.g. G.728, GSMFR, G.722) [4].

Actual packet switching voice systems include not only IP traffic capable phone end nodes, but application servers responsible for signaling and cost accounting as well *(Figure 1)*. Signaling methods (like SSCP, SIP, etc.) in this environment are much more intelligent than those used in PSTN networks (e.g. QSIG). The voice content transmission is realized by RTP (Real Time Protocol) directly between IP phone nodes. Signaling is transmitted in TCP, and digitized voice is transmitted in UDP segments.

To interconnect VoIP and PSTN networks special gateways are used that are capable of converting signaling and voice content as well. The IP phone compiles messages from the sampled voice and assembles voice segments by a codec module *(Figure 2)*. Voice segments transmitted by RTP on top of UDP are shaped by jitter buffers at the receiving node. Modules like G.711, G.723, G.728, and GSM are called narrow-band codecs and utilize at most 64 kbps voice rate [4]. Codecs like Brand-Voice32, G.722, etc. need higher bandwidth for the increased voice quality.

The main characteristics of IP voice codes are: voice bit rate, length of voice frame (80-520 bytes), duration of voice frame (0.125-20 ms), the IP packet bandwidth (24-272 kbps) and the delay of voice transfer (0.25-40 ms).

Figure 1. IP phone and VoIP architecture

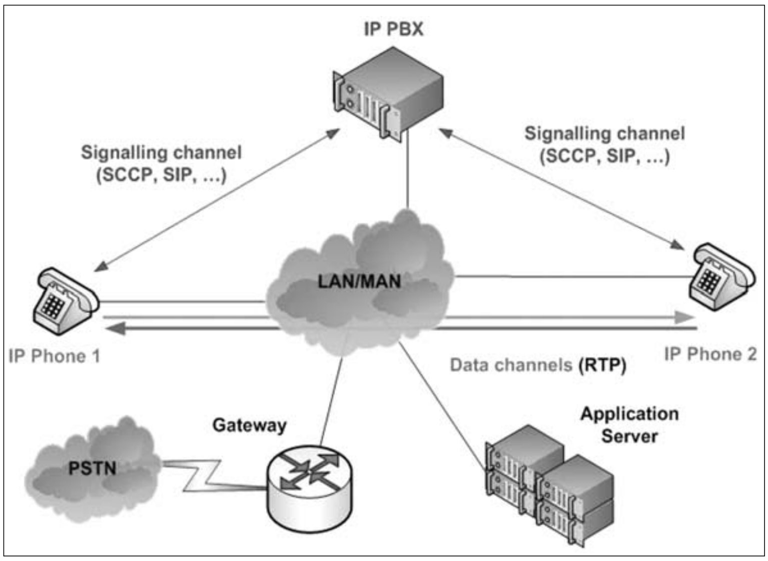
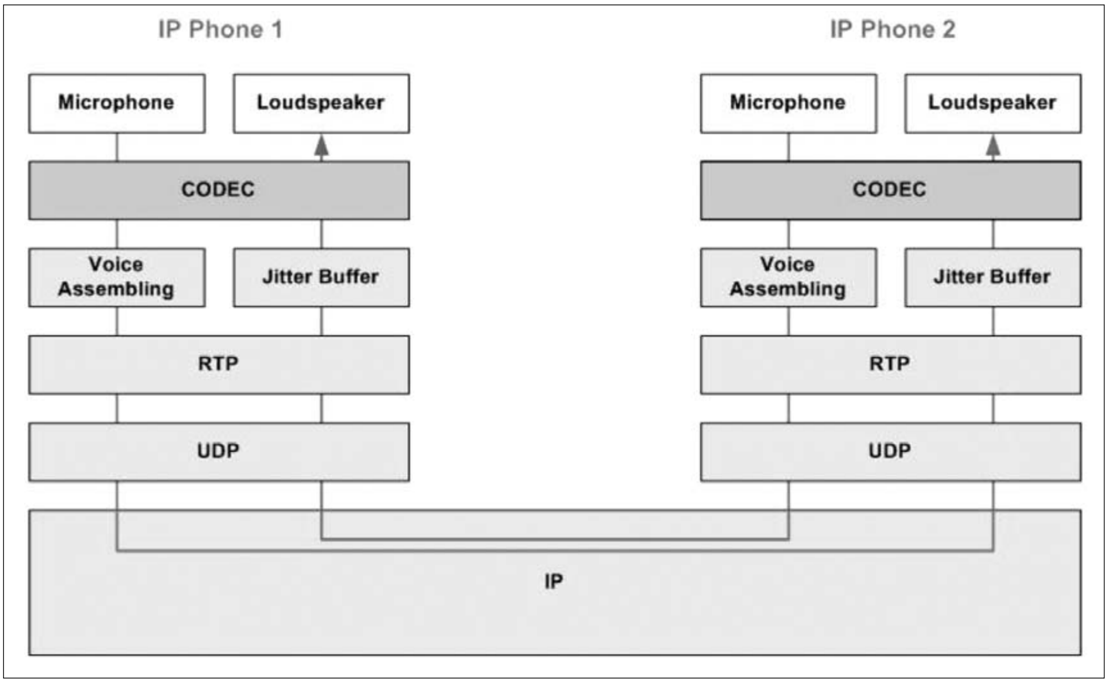


Figure 2.
VoIP transmission model



These parameters depend on the codec type. The delay is influenced by eight mechanisms: sample buffering, coding, packet assembling, transmitting, transferring through LAN/MAN, receive buffering, decoding and playing back. To guarantee interactivity, this delay can be at most 200 ms (high voice quality) and 400 ms (eligible quality) respectively.

Different applications need distinct requirements regarding transferring the traffic through the LAN/MAN. The necessary network resource is a function of the traffic volume generated. Some applications are more tolerant of the transfer delay and delay variation, but others cannot tolerate a given transmission error limit excess. Basic QoS parameters are bandwidth, latency, jitter and transmission error rate [4]. The QoS algorithms manage the network resource utilization: dedicated bandwidth allocation, monitoring the transmission error characteristics, congestion avoidance and management, traffic shaping, traffic prioritization.

The three well known QoS mechanisms are: *Best-Effort, Intserv* and *Diffserv*. Intserv is based on IETF RSVP (Resource ReserVation Protocol) for QoS between end user nodes. Diffserv works among intermediate nodes on L2-L7 layers of OSI. The most important feature of "coloring" the IP traffic is the DSCP (Diffserv Code Point) field in the IP packet header. The ingress interface needs classification, marking, policing, shaping (e.g. FIFO, FQ-Fair Queue, WFQ-Weighted Fair Queue, WRED-Weighted Random Early Detected, "tail-drop", LLQ-Low Latency Queuing), while the egress interface needs congestion avoidance, policing, and shaping tasks, respectively. The voice is classified as the traffic with the highest priority.

Modeling packet switched networks is mainly characterized by manipulating only the packet arriving time series as a stochastic process [8]. There are only few papers that study both the length and the arriving time of packets in evaluation of PDU real time transmission performance [9]. In this paper, we evaluate both arrival time and length of packets as time series for evaluating the performance of QoS mechanisms. Because the bandwidth of the wired channel between two intermediate nodes in most cases (i.e. IEEE 802.3 protocol family) is technology dependent and is fixed, the frame size in our case can be transformed linearly into time dimension. Utilizing a new transformation, called ON/(ON+OFF), two time series can be obtained: channel utilization and packet/channel intensity. These measures can be used favorably for complex evaluation of the different QoS mechanisms.

In Section 2, performance characteristics of the IP networks and a special measure based on entropy, called Corvil bandwidth, are presented. In Section 3, wavelet analysis of self similar processes is introduced. Sophisticated methods of entropy and wavelet analysis applied on traces of IP phones in congested and congestion-less environment will be presented in Section 4. In the concluding section we also outline potential future work.

2. Performance characteristics of IP networks, entropy, Corvil bandwidth

The performance of current IP network applications is influenced by three factors: *bandwidth, statistical multiplexing* and *QoS mechanisms* [2]. The task of measuring *bandwidth* is relatively simple because intermediate nodes (routers, switches) are able to store five minutes of average values based on SNMP MIB objects. The detected values provide a measure of traffic traversing the network, but do not calculate exactly the necessary bandwidth for different network applications. Packet discarding and jitter are strongly influenced by the behavior of traffic at the millisecond time scale. There is no detailed information to predict the throughput of applications. In case of medium sized networks VoIP applications tolerate several 10 milliseconds of jitter. Practical experience of given network applications resulted in some empirical rules.

For example, in Best-Effort IP services, only 60% of the five minutes periods with a load above 95% for the network resources is recommended. If the load intensity exceeds this 60% threshold, the infrastructure needs to be improved. The percentages above are empirical and cannot be generalized for any network services. In VoIP environment the 60% load intensity referred above should be decreased to 40%.

An important tool of service providers is the *gain of statistical multiplexing,* which shares the resources of the packet switched networks randomly. When transmitting ten pieces of different video streams with the same characteristics on circuit switched network exactly ten times the bandwidth of a single stream is needed. In the case a packet switched environment significantly less aggregated bandwidth is needed. The reason is that the asynchronity of short bursts of the different streams making aggregated traffic is more shaped. The difference between the sum of individual bandwidths and the effective aggregated bandwidth is called gain of statistical multiplexing and is a measure of IP network performance characteristics.

Traffic shaping, policing and differentiated queuing are the most efficient *QoS mechanisms*. Dimensioning the bandwidth for robust statistical reliability, guaranteeing the prescribed gain of statistical multiplexing, and setting the QoS mechanisms can be managed in two ways: the first method is realization of highest gain with the requested quality guarantees, and the second method is based on extra dimensioning of network resources to produce performance. In the first case special services cannot be assured, while expensive network infrastructure is needed in the second case.

The uncertainty feature of network traffic is observed by the non deterministic relation among these three factors [3]. The network bandwidth, the traffic load, and the QoS objectives are essentially inter-related. Modification of either of them, influence the relation between others. The bandwidth necessary for a guaranteed delay depends not only on the network load, but also on the type of traffic (VoIP, data). The following set of formula shows the relationship among quality, network and traffic:

$$\left.\begin{aligned}\text{Quality} &= f_Q(\text{Network, Traffic})\\ \text{Network} &= f_N(\text{Traffic, Quality})\\ \text{Traffic} &= f_T(\text{Quality, Network})\end{aligned}\right\} \tag{2.1}$$

The CB (Corvil Bandwidth) technology gives proprietary solution to the equation system above in a given network environment. The method is based on intensive sampling of the traffic to extrapolate rules for dimensioning the network resources. The bandwidth is the simplest metric and can be measured relatively easily. Today SLAs (Service Level Agreement) specify the acceptable packet discard rate and delay parameters with SNMP, which are evaluated on weekly or monthly time scales.

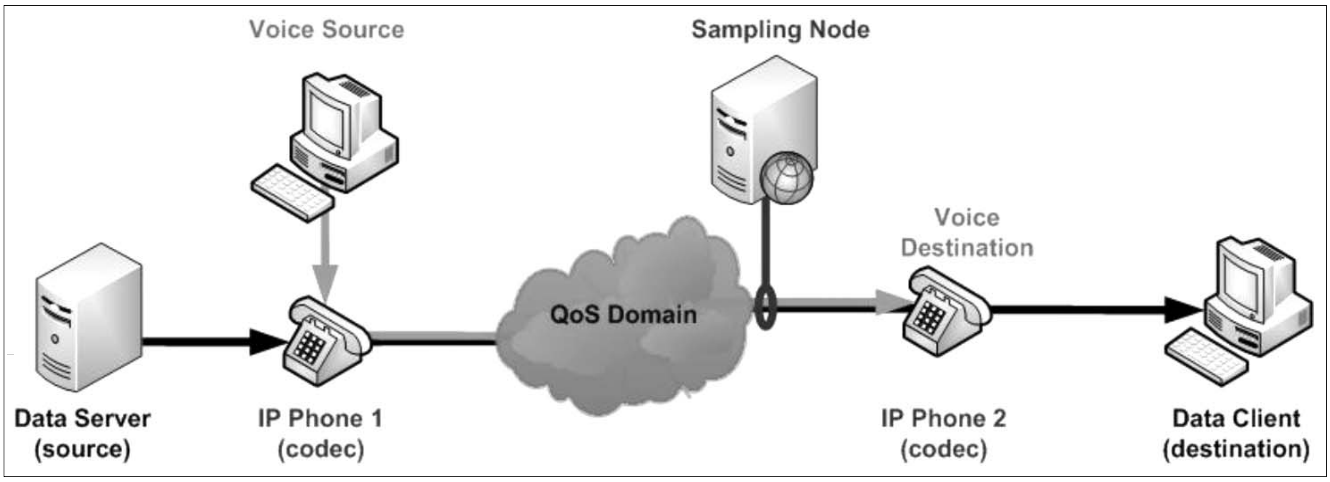
These are too rough values to assure guarantees for special or real time applications. The CB technology is based on Large Deviation theory applied to queuing systems, which analyzes the entropy, being the key statistical characteristic of a given queuing system. The entropy of a packet trace describes queue management and multiplexing of different traffics inside of network devices.

$$\text{CB} = f_Q(\text{Corvil Entropy, QoS}) \tag{2.2}$$

Some CB dimensioning rules of traffic classes or interfaces are the following: the queue processing delay is 0.001...1 sec and the size threshold is 1...2000 packets; the percentage of protected packets is 1...100% in steps of 0.0001%; the periods of protection rules are: 5 minutes, 1/2/4/ hours, 1 day, 1 week. The CB measured for 5 minutes differs strongly from the bandwidth measured by SNMP because CB considers sub-millisecond features of traffic. The real bandwidth need is set as a function of transfer delay and packet discard rate.

All the considerations regarding entropy suggest statistical analysis be effectuated at sub-millisecond time scales.

Figure 3. Measurement environment of the VoIP traffic



3. Wavelet analysis of self similar processes

A real valued $\{Y(t), t \in R\}$ process is self similar (H-ss) with Hurst parameter (H> 0), if for $\forall\, a > 0$ exists: $Y(at) \stackrel{\text{def}}{=} a^H Y(t)$ [8]. A real valued $\{Y(t), t \in \boldsymbol{R}\}$ process is H-sssi, if it is H-ss with stationary increments. If $\{Y(t)\}$ H-sssi has finite deviation, then $0 < H \leq 1$. A discrete increment time series can be created by $X_k = Y(k) - Y(k-1), k = 1,2,\ldots$. Let $X^{(m)}$ and $r^{(m)}(\cdot)$ be the m-aggregated time series and the autocorrelation function of X, respectively, where $X_k^{(m)} = \frac{1}{m}\sum_{i=(k-1)m+1}^{km} X_i$. In case of $0 < H < 0.5$ the process is called short range dependent (SRD), and for $0.5 < H < 1$ the process is long range dependent (LRD). If the process is LRD, then the form of autocorrelation function of the increment process is: $r(k) = \frac{1}{2}[(k+1)^{2H} - 2k^{2H} + (k-1)^{2H}]$. For exactly self similar process the deviation of aggregated increment process is $var(X^{(m)}) = m^{2H-2}var(X)$, and $r^{(m)}(k) = r(k)$. It can be observed that for LRD $var(X^{(m)}) > m^{-1}var(X)$, and for SRD $var(X^{(m)}) < m^{-1}var(X)$. Process X is asymptotically self similar, if for k high enough, holds: $\lim_{m\to\infty} r^{(m)}(k) = r(k)$ [7].

The discrete wavelet transformation (DTW) is a time-frequency decomposition, which assigns two-variable coefficients to the time series X with n elements, in the following formula [1]:

$$d_{j,k} = \int X(s)\psi_{j,k}(s)ds\,, j \in \boldsymbol{Z}, k \in \boldsymbol{Z} \tag{3.1}$$

where any wavelet has the following unique expression:

$$\psi_{j,k}(s)=2^{-j/2}\psi(2^{-j}t - k) \tag{3.2}$$

Several mother wavelets exist and each has the following features:

$$\int t^k \psi(t)dt \equiv 0,\ \forall\, k = 1,2,\ldots,N-1 \tag{3.3}$$

The wavelet decomposition is a linear combination of mother wavelet functions and coefficients $d_{j,k}$:

$$X(t) = \sum_{j\in\boldsymbol{Z}}\sum_{k\in\boldsymbol{Z}} d_{j,k}\psi_{j,k}(t) \tag{3.4}$$

The wavelet coefficients can be used to evaluate scale, and frequency dependence features of LRD processes. The second order log-scale diagram (2-LD) is a log-linear graph of the estimated second moment dependent of the octave j:

$$\mu_j = \frac{1}{n_j}\sum_{k=1}^{n_j}\left|d_{j,k}\right|^2 \approx 2^{j(2H-1)}, \text{ where } n_j = 2^{-j}n \tag{3.5}$$

The average of square sum of the wavelet coefficients is called energy function, μ_j of the time series X. Conforming to (3.5), the logarithm of the energy function is a linear function of octave j.

$$y_j = log_2(\mu_j) \approx (2H-1)j + c \tag{3.6}$$

For the estimation of the Hurst parameter linear segment or segments of 2-LD can be utilized. If more than one linear segment can be identified, the process is multifractal, otherwise is monofractal. The H parameter for the linear octave segment $[j_1,j_2]$ can be evaluated by the WLS (Weighted Less Squares) method with the following formula [1]:

$$\hat{H}(j_1,j_2) = \frac{1}{2}\left|\frac{\sum_{j=j_1}^{j2} S_j j y_j - \sum_{j=j_1}^{j2} S_j j \sum_{j=j_1}^{j2} S_j y_j}{\sum_{j=j_1}^{j2} S_j \sum_{j=j_1}^{j2} S_j j^2 - \left(\sum_{j=j_1}^{j2} S_j j\right)^2} + 1\right|, \quad \textit{where } S_j = \frac{n\, ln^2 2}{2^{j+1}} \textit{ are the weights.} \tag{3.7}$$

4. Measurement environment and analysis of the measured processes

a) Analysis of VoIP traffics in congested environment

The link between the source and destination is 10 Mbps Ethernet, on which (T) TCP and (U) UDP traffics were generated at the maximum load of the channel with Java based IPerf server and client entities running on the Data Server and the Data Client, respectively. Both voice and data traffic were transmitted through the LAN interface of the IP phones *(Figure 3)*.

One minute long songs, (H) hard rock (Limp Bizkit – Eat You Alive) and (P) piano (Wolfgang Amadeus Mozart – Concert for horn and orchestra KV KV 285d C major Adagio non troppo) were repeated on the source and transmitted from the IP Phone_1 to IP Phone_2. Different types of codecs (G.728, GSM, G.711, WideBand-G.722) were utilized at the IP phones, while the voice traffic inside of QoS LAN/MAN domain was regulated by DSCP values set for 0x00-"best-effort", 0x02-low price, 0x04-reliable, 0x08-performance, 0x10-low latency. The eighty different traffic traces were created by varying the transport protocol, the song type, the codec type, and the DSCP value: [(T,U) x (H,P) x (G.728,GSM,G.711,WB) x (0,2,4,8,16)] = 2x2x4x5 = 80 traces. These traces were captured by the program Wireshark with 1 μsec accuracy.

b) Analysis of VoIP trunk traffics in congestion-less environment

The aggregated traffic on the voice VLAN of IP/PBX gateway was captured for a population of 1500 IP phones [6]. The voice trunk link was 100 Mbps Ethernet and the capturing task was effectuated with 1 μsec accuracy in university environment on a working day for a one hour time interval.

For both a) and b) scenarios arrival time and the length of the Ethernet frames were captured. In case of voice transmission the Ethernet MTU (1500 B) is at least two times higher than the VoIP packet size, so there were no packet fragmentations. Because Ethernet is the mostly applied technology in the access and the distribution network layers today, the measurement scenario and the packet traffic behaviors can be considered for LAN and MAN environment as well. The 1 μsec accuracy of the L2 PDU capturing is caused by the Corvil bandwidth considerations in Section 2.

Metrics studied in network traffic analysis are related mostly to the effect of the network resources utilization (e.g. frame inter-arrival time, frame size). Lots of dominant papers discuss statistical features (LRD, fractal, multifractal) of packet switching based on measures [9,7]. A new transformation, called ON/(ON+OFF) is proposed and introduced in this paper, which offers load evaluation of the network resources directly. Our implication is that analyzing the finite set of network resources (channel load, channel activation intensity) provides more suggestive measures of the communication processes and gives more clear view of the instantaneous network resources state.

The length of the frame given in bits can be converted in time dimension, making it possible to calculate two time series L_i, the average transmission time interval (ON_i), and $tan(\varphi_i)$, the channel load. The evaluation time interval $T = M * (T_{ON} + T_{OFF})$ was fixed, and for each measurement, $T = 100\ ms$. M_i is the intensity of frame arriving and can be considered as the intensity of channel activation in the evaluation time period i. The instantaneous channel load and the phase of voice traffic for evaluation period i can be calculated with the formulae (4.1). Basic time diagrams of the ON/(ON+OFF) transformation are presented in *Figure 4*.

$$\left.\begin{array}{ll} M_i\ [\,], & \textit{Intensity of channel activation} \\ L_i[sec], & \text{Average } \textit{frame processing time} \\ D_i = \sqrt{L_i^2 + T^2}\ [\text{sec}], & \textit{Square average time} \\ \tan(\varphi_i)\ [\%] = \frac{L_i}{T} * 100, & \text{Average } \textit{channel load} \\ \varphi_i[Rad] = \tan^{-1}(\frac{L_i}{T}), & \text{Average } \textit{channel phase} \end{array}\right\} \quad (4.1)$$

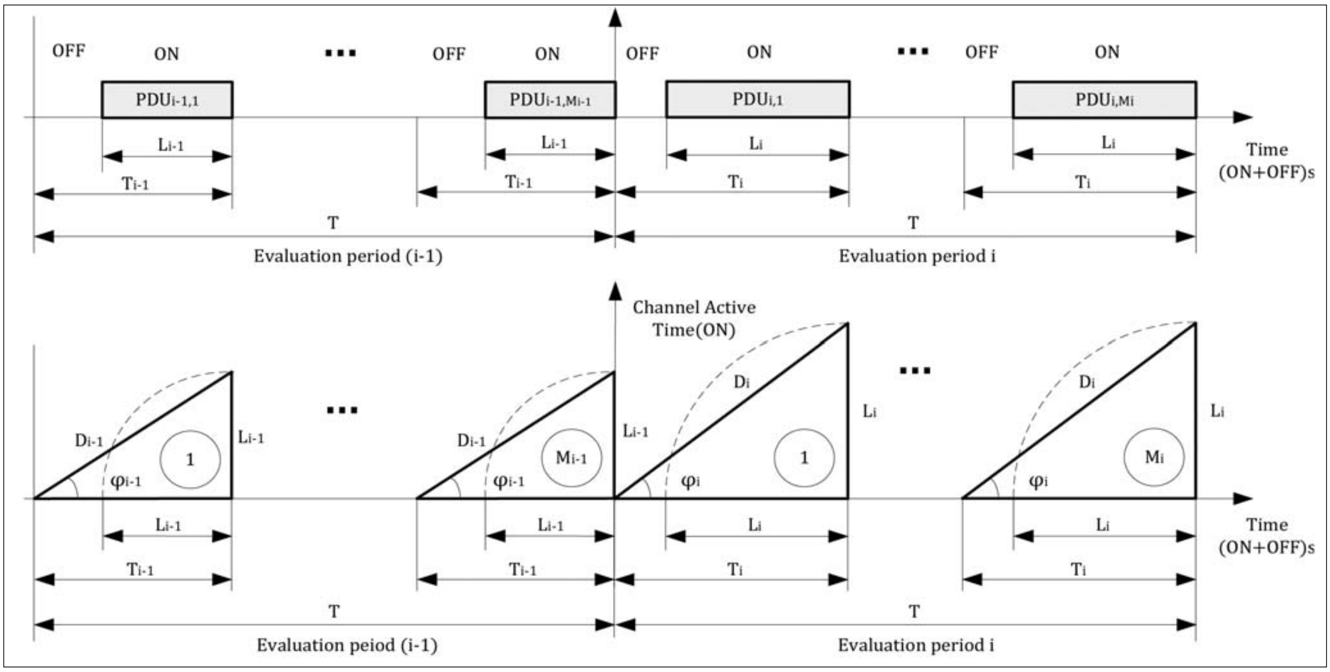
For VoIP congested environment the main characteristics of the four types of codecs are presented in *Figure 5*, and the relative deviation of the eighty intensity series can be seen in *Figure 6*. The relative deviation of the channel load and the intensity for voice are practically the same. In Figure 6, the dark colors mean smaller values, and the light colors mean larger values.

In case of DSCP=0 ("best effort"), TCP data traffic and G.711 voice codec, the relative deviation of voice traffic intensity is small, but for other types of codecs this can be as high as 20% (for GSM). When the voice traffic is treated with QoS mechanisms, this relative deviation remains at low values. In case of UDP data traffic the relative deviations of voice loads are high, but with TCP data traffic these are small. This phenomenon is caused by the TCP flow control mechanisms, which decrease and shape the TCP data traffic in favor of the UDP voice traffic. For UDP data traffic there is no flow control, and the voice traffics without QoS have larger deviations. The dynamics of songs has an effect on the voice traffic load only for low bit rate codecs (G.728, GSM).

Figures 7–10 present the channel load time series and their wavelet transforms in cases of UDP data traffic, GSM codec, "best-effort"/QoS and piano/hard rock song environments. Although the characteristics of the two time series are comparable, the main difference is shown expressively by the wavelet transforms.

Table 1 and *Figure 11* show the estimated Hurst parameter, ($\hat{H}$) of the channel load and of the intensity for all eighty traces. Irrespective of the data flow transport protocols, in cases of G.728 and GSM codecs, the channel load time series are not self similar, why the $\hat{H}$ >1. In contrary, for G.711 and G.722 (WB) codecs the voice time series are self similar and LRD. Irrespective of the song dynamics and data flow transport protocols, in cas-

Figure 4. Basics of the ON/(ON+OFF) transformation



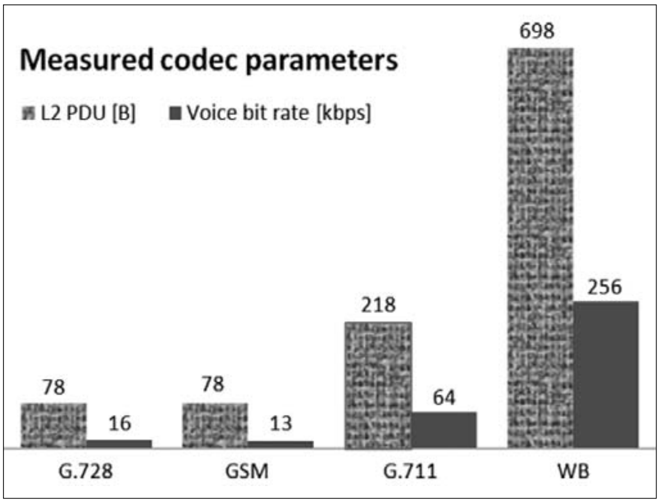


Figure 5. Characteristics of voice codecs

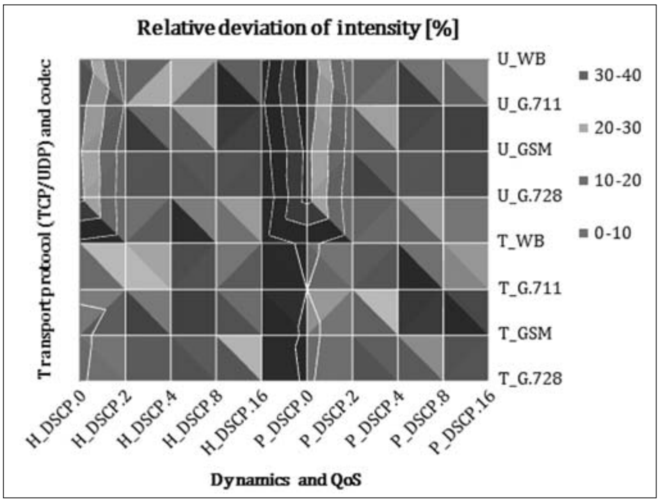
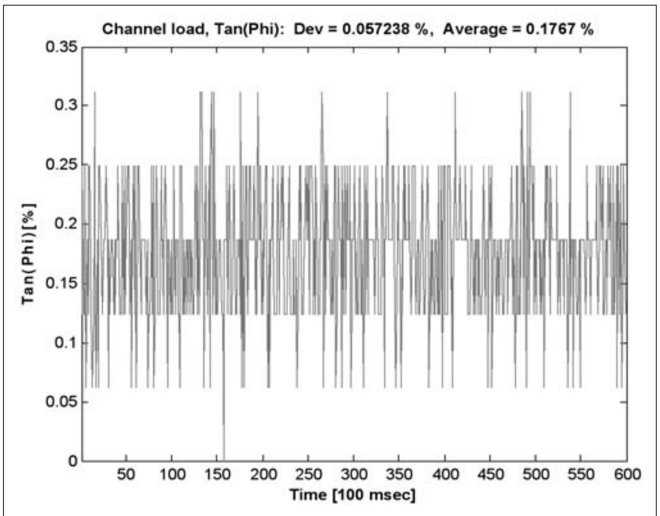


Figure 6. Intensity, M

es without QoS (i.e. DSCP=0), the voice channel load is H-sssi and LRD, and for the estimated Hurst parameter, $\hat{H} \in [0.56, 0.91]$.

The estimated Hurst parameter, $\hat{H}$ of voice traffic load varies contrary with the bandwidth of voice codec (Figure 5 vs. Table 1). The experienced voice quality at the receiving IP phone was better for codecs with higher bandwidth, and the congestion of voice traffic was not subjectively detectable when QoS mechanisms were active.



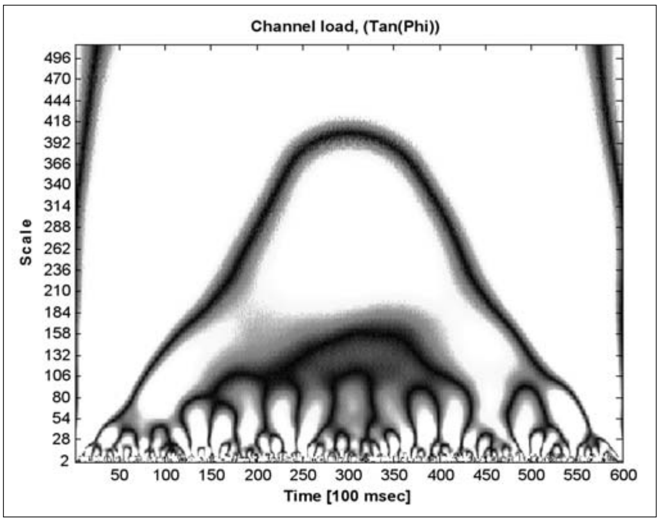
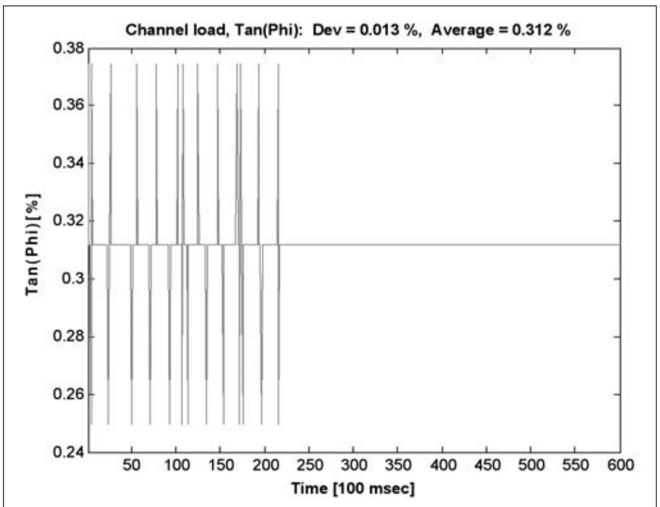
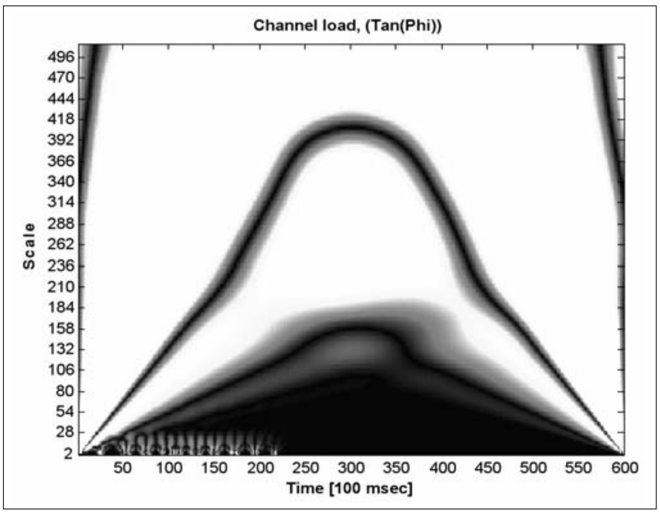


Figure 7. Channel load, Tan(φ) – UDP, GSM, no QoS, Piano – Figure 8. Wavelet transform, Tan(φ)

Figure 9. Channel load, Tan(φ) – UDP, GSM, DSCP=16, Hard Rock – Figure 10. Wavelet transform, Tan(φ)

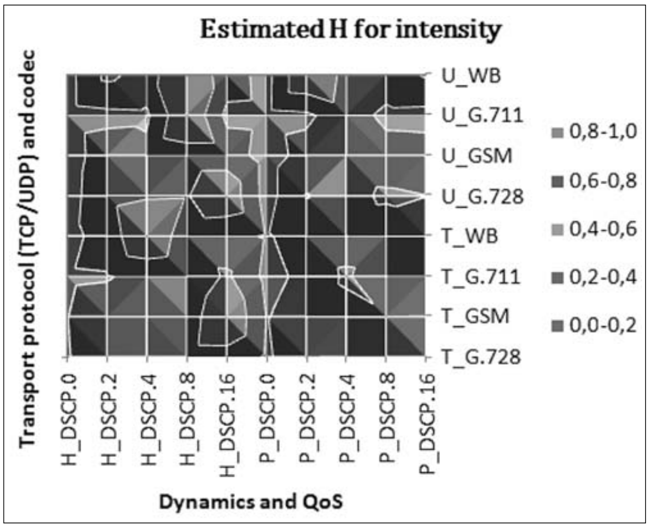




	T G.728	T GSM	T G.711	T WB	U G.728	U GSM	U G.711	U WB
H_DSCP.0	0,86	0,87	0,79	0,59	0,92	0,91	0,73	0,57
H_DSCP.2	B	B	0,73	0,76	B	B	0,75	0,93
H_DSCP.4	B	B	0,83	B	B	B	0,80	0,81
H_DSCP.8	B	B	0,74	0,76	B	B	B	B
H_DSCP.16	B	B	B	0,74	B	B	0,77	0,79
P_DSCP.0	0,87	0,85	0,78	0,57	0,91	0,89	0,72	0,56
P_DSCP.2	B	B	0,83	0,77	B	B	0,80	B
P_DSCP.4	B	B	B	0,79	B	B	0,96	0,82
P_DSCP.8	B	B	0,75	0,77	B	B	0,78	0,81
P_DSCP.16	B	B	0,74	0,81	B	B	0,76	0,90

Table 1. Estimated Hurst parameter of Tan(φ)

Figure 11. Estimated Hurst parameter of M



The intensity time series are LRD and for each analyzed case $\hat{H} \in [0.52, 1]$. Irrespective of the data flow transport protocols, in cases without QoS the estimated Hurst parameter of the voice traffic intensity is higher but near 0.5, $\hat{H} \in [0.51, 0.6]$. In case of G.711 codec, the song with high dynamics and *DSCP* = 8, performance optimization QoS mechanism causes high value for estimated Hurst parameter of voice traffic intensity. The intensity of this type of song traffics produces higher $\hat{H}$, than songs with low dynamics (Figure 11.)

For congestion-less VoIP environment, *Figures 12-13* present the channel load and the intensity of voice trunk, *Figures 14-15* present the 2-LD charts of them, according to relation (3.7). Although the shifting averages of the graphs indicate correlation, the characteristics of these two time series differ significantly because of the local maximums of the intensity. The 1 second length shifting average function of the channel load indicates the number of simultaneous voice sessions. The relative deviation of the channel load is 53%, and of the channel intensity is just 44%. The VoIP trunk traffic in congestion-less environment exhibits a multifractal feature as well.

The wavelet estimation of Hurst parameter of the channel load is $\hat{H} = 0.88$ and is less scale dependent, but for the intensity time series $\hat{H} = 0.61$ and for larger octaves is radically varying (Figures 14-15). The Ethernet link is congestion-less because of the small channel load, and the aggregated voice traffic is self similar and LRD.

5. Conclusions and future works

Voice transfer over IP networks is the most critical real-time network application and providing this service is a complex task and for service providers in IP LAN/MAN

Figure 12. VoIP trunk channel load

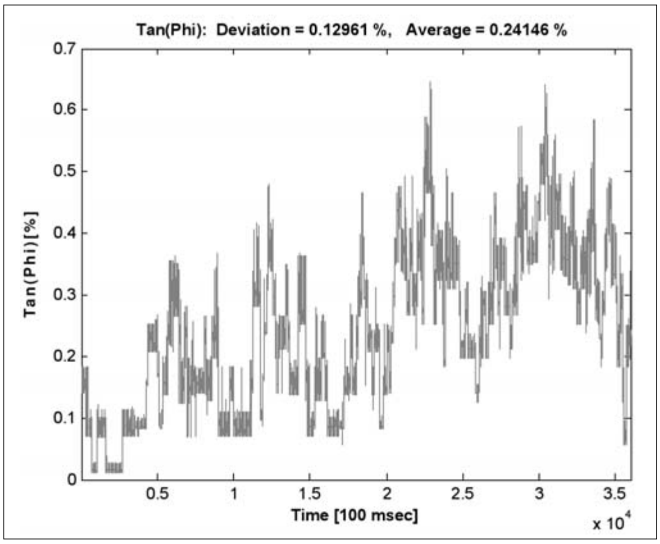
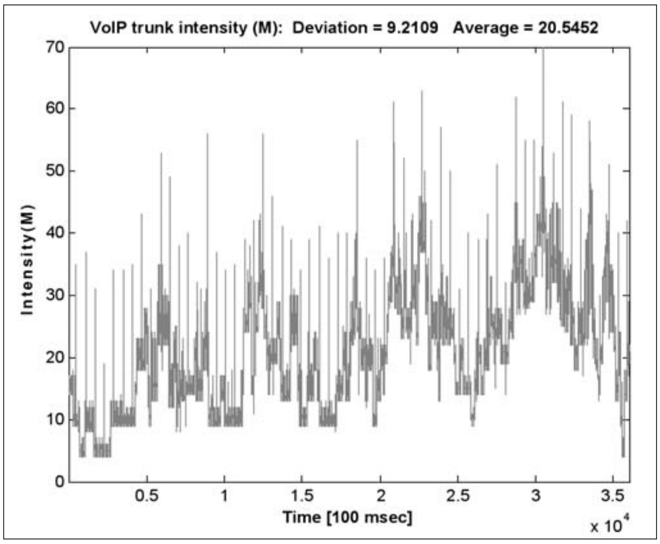
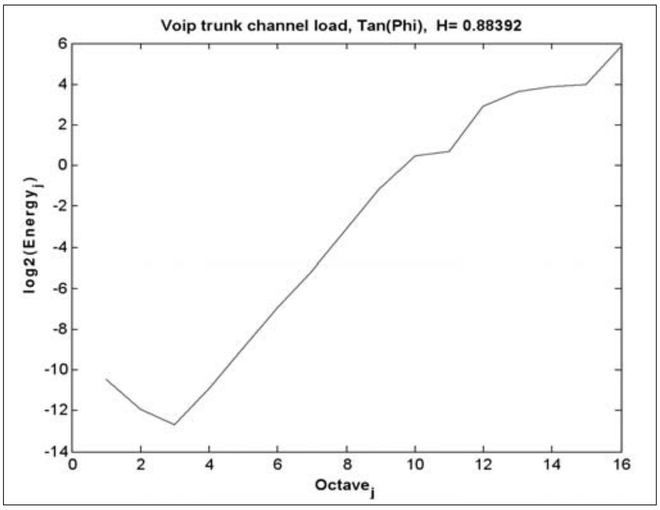


Figure 13. VoIP trunk channel intensity





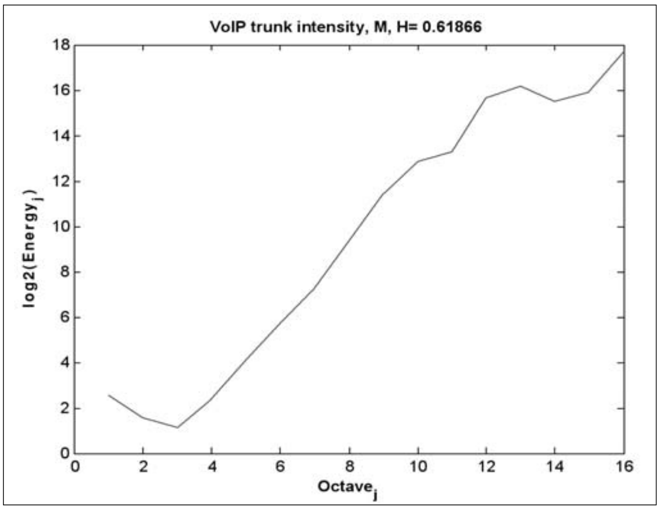


Figure 14. VoIP trunk channel load – H estimated ($\hat{H}$) with wavelet method – Figure 15. VoIP trunk intensity

environment it could be more difficult task than real-time video transmission service. The baseline QoS traffic classes based on the packet loss concealment threshold gives this priority order because only 20 ms are allowed for VoIP and higher values are permitted for real time video. The type of the codec determines the quality of the voice channel. The VoIP channel works on UDP transmission protocol, where there is no feedback, no flow control and no voice frame length modification during the session. The measured and subjectively sensed quality of the codec depends of both the physical channel bit rate and the packet switched protocol data unit size. In case of analyzed scenarios the increasing order of voice codec quality is: G.728, GSM, G.711, WB (G.722). This order was detected by independent human users during the measurements as well. The fractal phenomenon of L2/L3 voice traffic in congested LAN/MAN environment produces significant degradation of the voice quality. Wavelet analysis is an efficient tool for evaluating the H parameter, the fractal and scale dependent features of packet switched voice traffics, together with offering suggestive classification method of QoS controlled packet traffics. Based on the proposed ON/(ON+OFF) transformation VoIP network operators can determine more exactly the necessary NGN resources for a given number of voice subscribers not only in best-effort, but in congested and QoS controlled wired LAN/MAN environment, too.

QoS is becoming a service in the control plane of modern network protocols and strongly influences the self similar and fractal nature of transmission processes of the packet switched protocol data units. The quality of voice transmissions, the self similar and LRD features of these traffics are greatly influenced by the QoS mechanisms. This impact opens new directions in the area of QoS based flow control research area. More complex analysis is needed for employing both the channel load and intensity metrics of the IP packet switched protocol data unit traffic simultaneously. Based on the results of deep traffic research, optimal configuration profiles can be set for intermediate nodes to provide the necessary NGN QoS services. For this reason measurements and statistical analysis of communication processes are required at 10...100 μs time scale to cover the Corvil bandwidth based entropy characteristics and its macro effects during the session time of real-time applications.

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