Statistical aspects of network based real-time group communication and collaboration services

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Abstract

Deep studies of available real-time Group Communication and Collaboration (GCC) solutions such as voice, video and data teamwork tools have been performed by several international projects. The main goal of these works is to compare the applicability of these tools in the production job environment, and give recommendations on how such services can be provided to these groups. The total bit rate allowed per client is 256–2048 kbps for current GCC tools. This amount includes audio, video, control, web content and whiteboard traffic as well. The IEEE 802 type LAN/MAN communication technologies with best effort characteristics need quality of service (QoS) guarantees to provide real time service to the multimedia applications. The H.323/H.264 and the H.261 network protocols use relatively high number of UDP ports to accommodate automatically to the network layer comportment deeply influenced by bursts. This aspect explains the reduced number of service class configuration possibilities at the GCC client software. Even SOHO users with 256-512 kbps ADSL access need connection to the GCC servers. Analysis of mechanisms influencing traffic QoS is required. In the paper statistical evaluation of network traffic of the group communication and collaboration applications are effectuated with special interest regarding efficient regulation possibilities of the transmission mechanisms at low bit rate. Influence of the regulation will be presented based on time series captured in real multimedia network environment.

Keywords: TCP/IP, H.323, H.264, H.261, GCC service, best effort, burst, SOHO, ADSL, self-similarity, Hurst parameter.

1. Overview of the real-time GCC services

Group collaboration is a hot topic in the IT industry. Updates on existing GCCS systems and integration of different systems become available every day. Even more advanced systems are being developed by the software houses. Several international projects study available group collaboration solutions such as voice, video and data collaboration tools and give recommendations on how such services can be provided to public sector, including how to (re)use existing services available in the large LAN/MAN environments. Extensive analysis on all available real-time GCC services for those applications are given in the literature [1]. There is a legacy consisting of participants that already have conferencing and/or collaborating hardware and software, and they want to be able to keep on using it.

Supporting standards and interoperable systems is essential. In most cases on the network side is expected that all participants have a high bandwidth connection offering at least several megabits per second. In practice this assumption is not materialized, consequently need other studies for bit rate less than 2 Mbps to set up connection with QoS among end nodes and GCCS servers. Whereas recommendations regarding the underlying network infrastructure is weakly supported. Network problems are supposed to be dealt with by the service provider.

In most educational scenarios audio is key, video is sometimes required. However, there will also be scenarios where video quality is very important. Most researchers are very experienced email users and share documents by email, take them to the meeting and produce their documents with. Application sharing is required also by the users. Usually this means advanced whiteboarding or the sharing of dedicated applications that are not widely available to every participant. Collaborative document editing is considered a welcome feature. Audio should be comparable to the telephone system and is key, followed by the requirement for data sharing. Using video is an option in most scenarios. In some scenarios however, video is important and needs to be close to TV quality.

The environments in which the systems are used are universities, research institutes and offices of national research network organizations. Compared to business environments they have many more different kinds of PCs and workstations and operation systems. Collaboration in these environments is a multiple problem, meaning there is no one administrative domain all users belong to. They are users from different organizations, connected by different networks. The only common characteristic is access to an IP network. This leads to the requirement to use interoperable software and hardware working over different domains.

2. Basic functions of the tested GCC System

Several types of meetings the users could think about. There are a number of basic functions as a starting point to make a first choice of a supporting system. This list has a number of variables, like the size of the meeting, e.g. up to 20 experienced users versus big audiences; available equipment/systems, e.g. hardware codec versus

phone; organization, of the one or more speakers, who should interact, etc. There are four sets of meetings. a.) Project meetings include phone/videoconference meeting, documents on one web server and mail. b.) Meetings with simultaneous editing can be realized by H.323 videoconference, documents on one web server, mail, web based collaboration, some H.323/SIP clients connected through MCU. c.) In case of one way presentations (1 speaker, large audience) the server is needed to stream to large audiences. The feed is coming from a media encoder and needs player for viewers, VNC (audio/video + documents or applications) server for speaker, web based collaboration with no feedback allowed. d.) Presentations with feedback (multiple speakers, large audience) use media encoder/server, VNC (audio/video + documents) and web based collaboration tool.

Two departments of the University of Debrecen, Faculty of Informatics and Service Center for Informatics took part with 20 minutes live video presentation on the H.323 based Megaconference VI, organized by the Ohio State University in December 2004. The subject of the two-sited interactive presentation was Life at the Eastern part of EU. The videoconference was managed with two voice activated conference room devices. Based on the Megaconference sponsors opinion, University of Debrecen won Marratech GCCS server license for two months as a prize. One year later University of Debrecen bought Marratech server license for 50 concurrent users and unlimited virtual rooms.

This system is real-time desktop web collaboration an communication application with videoconference, phone, presence, chat, presenting, whiteboard, application sharing and session streaming possibilities. The server can be set up for any number of concurrent virtual rooms. Each room has following parameters: dedicated bit rate for audio, video data, video control, web slides, reliable whiteboard, best-effort whiteboard respectively, and type of codec (H.261 or H.264). The client software can connect to the server with three different video qualities (high, medium, low).

3. Measurement environment, measured values

Two multimedia desktop nodes $Client_A$, $Client_B$, one high performance server with 6000 MIPS throughput and one capture desktop were connected to a high speed (1 Gigabit/sec) fully switched Ethernet LAN. $Client_A$ and $Client_B$ were running Marratech client software with full functionality and node Server was the GCCS provider (see Figure 1). The Capture node was running TCPDump software for sampling the traffic at the Server network interface card. For snooping Server input and output dataflow LAN switches were configured to mirror the physical Ethernet switch port of the Server to the Capture node. All types of data flows between $Client_A$ and $Client_B$ were running through the Server during the measurements. Whereas snooping the Server Ethernet port all bidirectional data flows among the GCCS nodes were collected by node Capture.

Data flows between GCCS nodes are presented in Figure 2. There is no bottleneck inside of the LAN, because the physical channel is faster with two order

of magnitude relative to the analyzed multimedia traffic. Multimedia content is transmitted by UDP and control packets are transmitted by TCP protocol. Both, $Client_A$ and $Client_B$ send Real-time video and audio to the Server.
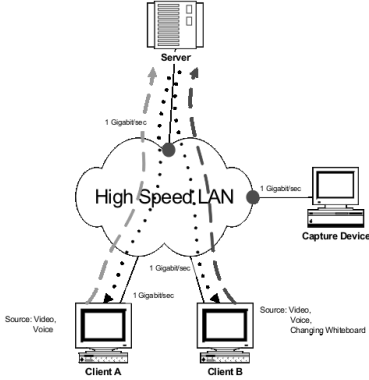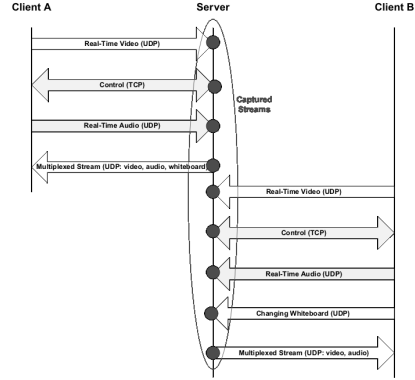

Figure 1: Measurement environment

Figure 2: Data flows

Control channel is bidirectional for client-server communication but the amount of signaling information transmitted periodically (10 seconds) is only several bytes. For this reason TCP flow is negligible compared with the UDP dataflows bit rate. Multimedia streams from the clients are not the same because $Client_B$ sends extra flow with content of the changing whiteboard. Both clients receive multimedia content through a multiplexed stream. $Client_B$ receives live audio and video form the Server, but $Client_A$ receives live video, audio as well and changing whiteboard content from the Server in addition. All data flows are transmitted in UDP packets.

There were running two multimedia content generators, one per client to transmit variable multimedia content during the sampling interval, named mmc_A (audio, video) and mmc_B (audio, video, changing whiteboard) from $Client_A$ and $Client_B$ to the Server, respectively. mmc_A is the same for each measurement task, and the same fact is valid for mmc_B as well. One measurement task (MT) is characterized by following quintuple:

$$MT_{ijk} = (mmc_A, mmc_B, M_i, C_j, Q_k), \quad i = 1, \ldots, 7; \quad j = 1, 2; \quad k = 1, \ldots, 3$$

where mmc_A, mmc_B are multimedia contents from $Client_A$, $Client_B$. M_i is the virtual room allocated total bandwidth, C_j is the virtual room/client video codec type, Q_k is the client video quality. The allocated total bandwidth to the virtual room is presented in Table 1 (WR - Whiteboard Reliable, WB - Whiteboard Best-effort). The name of the room includes the total bandwidth allocated in Kilobit/sec. The virtual room/client video codec type and the client video quality were
$(C_1, C_2) = (H.261, H.264)$
$(Q_1, Q_2, Q_3) = (Low, Medium, High)$ respectively.

There were executed 42 measurement tasks in total, with the sampling interval 50 seconds each. Depending on the allocated total bandwidth of the virtual room the number of Ethernet frames captured per measurement task was between $8.0e+3$

and 27.0e+3. For each captured L2 frame two parameters were registered: arrival time, and frame size.

Room	Allocated Bandwidth [Kbps]					
Name	Audio	Video	Control	Web slides	WR	WB
M_0256	32	128	10	66	64	64
M_0384	50	200	10	104	104	104
M_0512	64	290	10	128	128	128
M_0768	64	500	10	175	175	175
M_1024	64	756	10	175	175	175
M_1536	80	1026	20	400	400	400
M_2048	128	1390	20	500	500	500

Table 1: Virtual room total bandwidth allocation

Because the arrival moment and the size of the frames are stochastic processes, we prefer to use average bit rate and average frame size calculated for a number of T equally sized time intervals for each measurement task. This short time interval was set to 100 msec, resulting $T = 500$, the number of equally sized intervals for each measurement task MT_{ijk}. Having the originally captured two time series simple mathematical calculation can be used to obtain two other series for each MT_{ijk}. The analyzed two processes are:

$$\left.\begin{array}{ll} x_{ijk}(t) & - \; Bit\; Rate[bit/sec] \\ y_{ijk}(t) & - \; Data\; Link\; Frame\; Size[Byte] \end{array}\right\} \begin{array}{l} i = 1, \ldots, 7;\; j = 1, 2 \\ k = 1, \ldots, 3;\; t = 1, \ldots, T \end{array}$$

Processes $x_{ijk}(t)$ can be compared favorable because each has same number of elements (T). For processes $y_{ijk}(t)$ adequate comparison is possible as well.

4. Analysis of the network traffic processes

Having equal number of elements for each $x_{ijk}(t)$ and for each $y_{ijk}(t)$ processes, several characteristics of the different multimedia data flows can be considered. There were calculated characteristics of the processes given in Table 2. Definition of the metrics are well known and are given bellow.

	Analyzed Characteristics						
	Mean	STD	RSD	VMR	Range	SK	NH
Bit Rate	[bit/sec]	[bit/sec]	[%]	[bit/sec]	[bit/sec]	+	+
Frame Size	[Byte]	[Byte]	[%]	[Byte]	[Byte]	+	+

Table 2: Analyzed metrics

The Hurst parameter (H) is a measure of the second-order self similarity of process $Z = (Z(t) : t = 0, 1, 2, \ldots)$ with stationary covariance and variance VAR,

where m is the aggregation level. The background of the self-similarity in network traffic analysis is well explained in the literature [2, 3]. In our case aggregation is made for a range $[200\,\text{msec}, \dots, 10\,\text{sec}]$ $(m = 2, \dots, 100)$. Metrics and parameter H given bellow are calculated and compared for each process $x_{ijk}(t)$, $y_{ijk}(t)$.

Mean
$$\overline{Z} = \frac{\sum z(t)}{T}$$

Standard Dev.
$$STD(z(t)) = \sqrt{\frac{1}{T} \sum [z(t) - \overline{Z}]^2}$$

Realtive STD
$$RSD(z(t)) = \frac{STD(z(t))}{\overline{Z}}$$

Var.-to-Mean Rat.
$$VMR(z(t)) = \frac{STD^2(z(t))}{\overline{Z}}$$

Range
$$Range(z(t)) = Max(z(t)) - Min(z(t))$$

Skewnes
$$SK(z(t)) = \frac{\sqrt{T} \sum [z(t) - \overline{Z}]^3}{\left(\sum [z(t) - \overline{Z}]^2\right)^{3/2}}$$

Norm. Hist.
$$NH = Histogram(\frac{z(t)}{T})$$

Par. Hurst (H)
$$\begin{cases} H(m) = 1 - \frac{1}{2} \log_m \left(\frac{var(Z^{(m)})}{VAR} \right), \ m = 2, \dots, 10 \\ z^{(m)}(t) = \frac{1}{m}[z(tm - m + 1) + \cdots + z(tm)] \end{cases}$$

Based on the figures obtained by placing the given metric values of different processes to the same plot, we can derive few statements. Each graph on the horizontal co-ordinates use the total allocated bandwidth for virtual rooms.

a.) The codec H.261 is less bandwidth consuming than H.264 (see Figure 3). The mean bit rate for low allocated bandwidth rooms is linear with the bandwidth. The 768 Kbit/sec bandwidth is cut-off point for mean bit rate in case of H.261 codec.

b.) Codec H.261 uses shorter L2 frames than H.264 (see Figure 4).
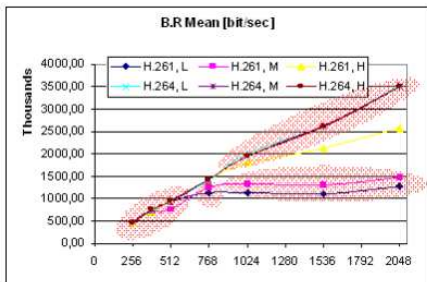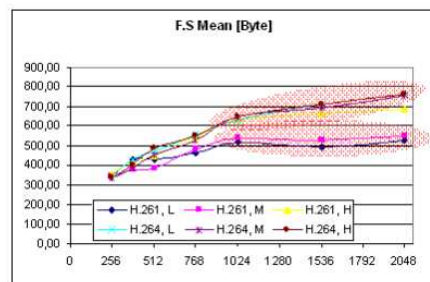

Figure 3: Bit Rate Mean

Figure 4: Frame Size Mean

c.) Codec H.264 is burstier than H.261, having higher STD for bit rates (see Figure 5). The standard deviation of the bit rate increases with the allocated bandwidth.

d.) For rooms with allocated bandwidth higher than 1024 Kbit/sec, the standard deviation of L2 frame sizes is practically constant and is equal to 100 Bytes (see Figure 6).
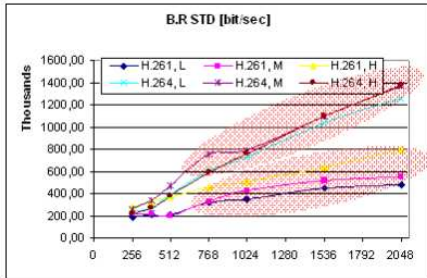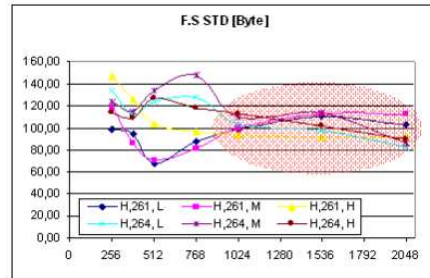

Figure 5: Bit Rate STD

Figure 6: Fame Size STD

e.) For room with allocated bandwidth higher than 1024 Kbit/sec the relative standard deviation is practically constant and is equal to 40% (see Figure 7). Rooms with lower allocated bandwidth are burstier.

f.) Low allocated bandwidth rooms use dispersed L2 frame sizes (see Figure 8).
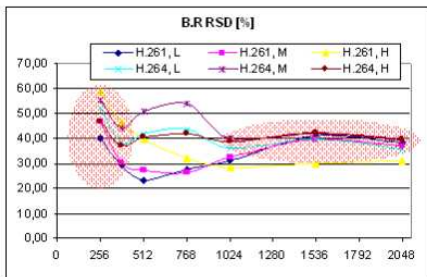
Figure 7: Bit Rate Relative STD
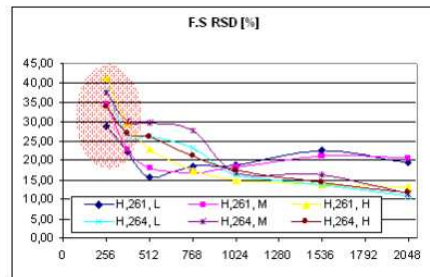
Figure 8: Frame Size Relative STD

g.) Codec H.264 is burstier than H.261, can be seen on Figure 9. Codec H.261 with reduced quality is best at the 512 Kbit/sec allocated bandwidth.

h.) The variance to mean ratio at the 1024 Kbit/sec allocated bandwidth is around 20 Bytes for each codec at any quality (see Figure 10).
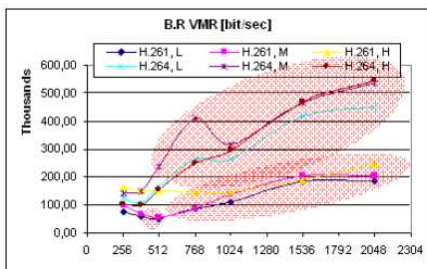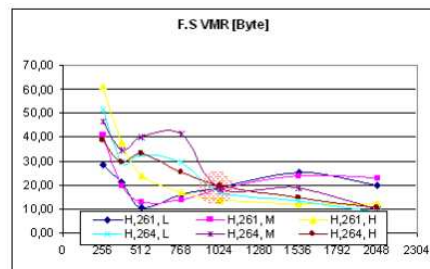

Figure 9: Bit Rate VMR

Figure 10: Frame Size VMR

i.) Codec H.264 needs higher bandwidth than H.261 (see Figure 11). The bit rate range is 10 Mbit/sec wide, even for 2 Mbit/sec multimedia sessions.

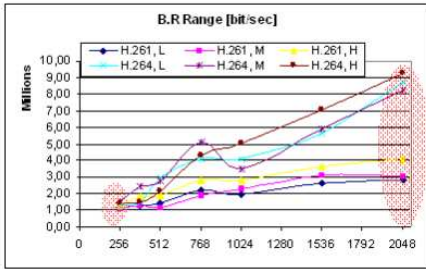j.) Codec H.264 with high quality needs 900 Bytes for 1024 Kbit/sec allocated room bandwidth (see Figure 12).
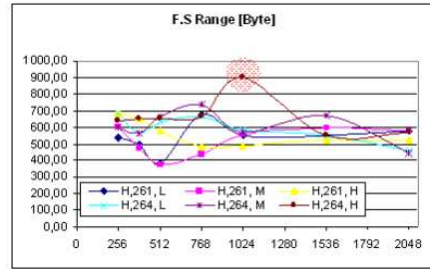

Figure 11: Bit Rate Range

Figure 12: Frame Size Range

k.) All bit rates are right-skewed (>0) (see Figure 13). This means that the bit rate is frequently higher than the average bit rate, the traffic is not bursty and there are relatively long inactive time intervals.
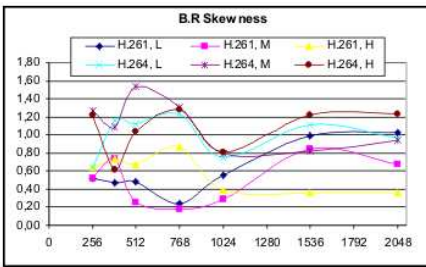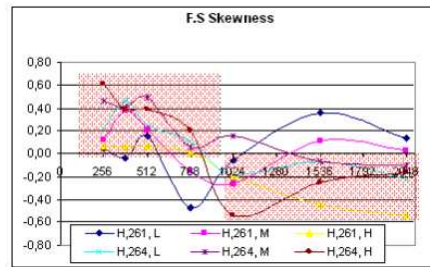

Figure 13: Bit Rate Skewness
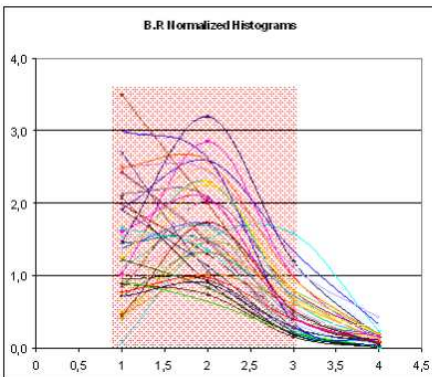

Figure 14: Frame Size Skewness
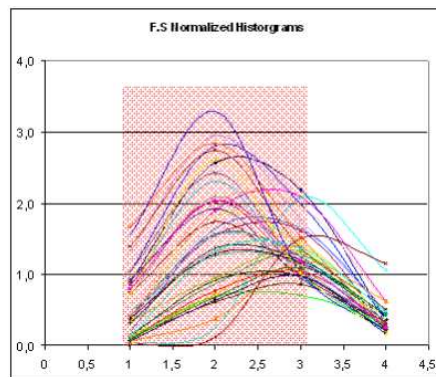

Figure 15: Bit Rate N. Histogram

Figure 16: Frame Size N. Histogram

l.) Rooms with allocated bandwidth less than 768 kbit/sec has frame sizes right-skewed (>0) (see Figure 14). Higher allocated bandwidth rooms has frame sizes

left-skewed (<0). The 768 Kbit/sec allocated bandwidth of virtual rooms is cut-off point for multimedia codecs.

m.) The bit rate interquartile range is 85% (see Figure 15).

n.) The frame size interquartile range is 60% (see Figure 16).

The calculated Hurst parameter for each process analyzed depends on the aggregation level, m and is a monotone descending curve. The range of parameter H for each process is [0.4, 0.95]. This phenomenon proves that the autocorrelation function can not be assumed slowly varying at infinity for UDP based multimedia traffic processes.

5. Conclusions and remarks

Reat-time group communication and collaboration data flows were captured and analyzed as time series for different allocated bandwidth and for two video codecs. There were compared basic statistical parameters. Codec H.261 (1990) has superior transfer quality than H.264 (2004) codec which is a contradiction outwardly. Codec H.264 is burstier than H.261 [4]. Low bandwidth rooms use high sized L2 frames and vice versa. GCCS needs MTU = 900 Bytes which is a special service criteria for SOHO/WAN technologies (ADSL, WiFi). GCCS need at least 5–10 Mbit/sec L2 bandwidth for good quality. MPEG-4 (H.264) codec is optimized for HDTV in LAN/MAN environment (>10 Mbit/sec). For SOHO/WAN environment with bandwidth less than 10 Mbit/sec H.261 has better characteristics than H.264. All these affirmations are useful for service providers and users as well. In the future efficient QoS mechanisms are required in low bit rate network environments. Needs profound analysis of real time UDP traffic in high speed LANs, too.

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