

Exact Time Speaking System

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Abstract — The paper describes a system for speaking the exact time, a VoIP solution produced by IRITEL, based on a family of domestic media gateways. System contains two media gateways for reliability reasons. Media gateway is a result of R&D effort in IRITEL. Exploitation experience and some traffic characteristics are described.

Keywords — RTP, service 95, SIP, VoIP.

I. INTRODUCTION

IN contemporary telecommunication networks TDM (*Time Division Multiplex*) technologies are rapidly replaced with IP (*Internet Protocol*) technologies which offer many advantages and contribute to emerging of NGN (*Next Generation Network*). In an effort to cope with this trend and to offer a domestic, local network adjusted solution, IRITEL has developed an *Exact Time Speaking system* – ETS as part of a family of IMG (*Integrated Media Gateway*) devices. Different variations of IMG devices can be used in access level of NGN or on a border between NGN and PSTN or PLMN networks.

ETS is working as a service 95 (exact time) of Telekom Serbia. It has replaced a system which had synchronization and interface problems. System is realized entirely in VoIP technology. It is connected to IMS (*IP Multimedia Subsystem*) of Telekom Serbia through SBC (*Session Border Controller*) over IP/MPLS network. ETS is doubled for reliability – it contains two IMG95 (*Integrated Media Gateway*) devices and two switches connected as in Fig. 1.

ETS is connected to two Telekom Serbia's networks – IP/MPLS for voice and signaling, and IT network for maintenance and synchronization with exact time servers (*ntp servers*). TDM subscribers access ETS through appropriate MGWs (*Media Gateways*) as ETS is in NGN domain. O&M functions are performed by computer NU which runs alarm and statistical functions, web interface, etc.

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IMG95 has 120 VoIP channels supporting G.711 and G.729 codecs, and a large number of VQE (*Voice Quality Enhancement*) functions.

The paper presents a description of ETS functions, benefits over the previous solution and first exploitation experiences.

II. ETS FUNCTIONS

The basic IMG95 function is to answer PSTN originated calls, over VoIP network, activate a voice machine which speaks the exact time twice, and disconnect the call. IMG95 voice machine contains all messages needed for service 95. These messages are generated, packetized and transferred to IP/MPLS network for every call separately. The call can be generated from PSTN or from NGN directly (SIP users or internet).

Calls are established using SIP [1]. Media (voice) is transferred using RTP/RTCP [2]. Synchronization with exact time servers is done using *ntp* protocol [3]. Calls are synchronized to time by a sound signal. When a subscriber dials 95, he hears at least one ringing tone, an introductory message, and a message: "It is exactly xx hours, yy minutes and zz seconds" which is emitted twice. Synchronization is done every 10 s using *ntp* servers of Telekom Serbia. Voice is packetized using codecs:

- *G.711 Appendix 1 and 2*: Linear PCM 64 kbps A law/ μ law, VAD/CNG and PLC,
- *G.729 A/B*: CS-ACELP 8 kbps.

For voice processing and quality enhancement the following techniques are used:

- *Echo cancellation*: according to ITU-T G.168-2000 ECAN 10 ms, 16 ms, 32 ms, 128 ms window,
- *Voice activity detection*: VAD – silence intervals are found by analyzing speaker activity,
- *Packet loss concealment*: PLC - ITU-T G.711 Appendix 1,
- *Comfort noise generation*: CNG - ITU-T G.711 Appendix 2,
- *Adaptive jitter buffer manager*: AJBM – jitter elimination.

ETS redundancy is established by connecting two IMG95 devices in an *active/hot-stand by* configuration. One IMG95 is active, and when found faulty, all connections are transferred to another IMG95. There are no traffic losses, except for connections already established. Switchover is activated by function CFNR (*Call Forward Not Ready*) in IMS.

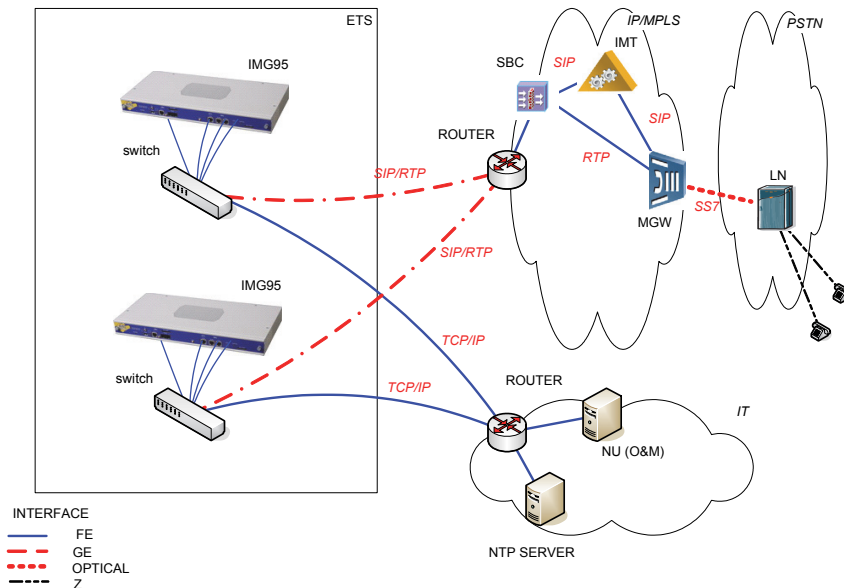


Fig. 1. Exact time speaking systems ETS.

ETS is surveyed from the distance by O&M computer NU. NU exchanges data with ETS using TCP/IP connection. Both IMG95 send data periodically regardless of NU state. Applications on NU are:

- Current or archived statistical review application,
- Configuration and alarms review application,
- Web application displaying state, configuration, user manuals, etc for IMG95 A and IMG95 B.

III. SYSTEM STRUCTURE

IMG95 can be divided into special entities – *engines* [4].

Communication between the engines is performed through a well defined mechanism – a *conduit*. Every engine can be realized on a separate hardware platform, which enhances the flexibility and scalability of IMG. The scheme of the structure is given in Fig. 2. The following engines exist in this scheme:

- SE *signaling engine*,
- AE *application engine*,
- ME *media engine*,
- TE *TDM engine*,
- VE *voice engine*.

SE establishes a signaling connection of IMG95 with IP world. SE can contain any of VoIP protocols (SIP, MGCP, MEGACO, etc.). IMG95 uses SIP. SE is realized on an IMG board.

AE performs multiple functions. It is the main part of IMG95 controlling all other applications, controlling ME and TE, communicating with SE and with outer world through O&M system. AE is realized on an IMG board.

ME turns TDM voice traffic into IP and vice versa using all the available VoIP mechanisms (codecs, quality control, echo cancellation...). ME is realized on a PFC board.

TE is an engine which prepares TDM channels for processing in IMG95. TE is directly connected with ME over TDM channels and with AE over a conduit. TE is realized on an IMG board.

VE generates voice messages in TDM channels and connects them with the basic board IMG over TE. VE is realized on VMU board.

A conduit is a mechanism connecting all engines. It is a software/hardware entity for message transfer. In IMG95 the conduit is based on TCP/IP protocol. SE, AE, ME and TE are connected over the conduit.

SE is connected using SIP protocol with IP/MPLS network.

AE is connected to IT network over TCP/IP.

ME is using RTP/RTCP protocol to connect to IP/MPLS network.

VE is connected with IT network over TCP/IP, and through this network with TE and AE.

TE is connected with VE over internal TDM multiplex and with ME over a H.110 bus.

Engines use numerous services performed in operating system Linux and its application layer.

IV. ETS HARDWARE

ETS hardware consists of three printed circuit boards:

- IMG – basic board,
- SPT8122PFC-HV (PFC) – board with DSP processors,
- VMU – voice machine unit board.

Board IMG is a control unit of IMG95. It contains: processor Power QUICC II, FLASH memory, SDRAM memory, etc.

PFC (MSC8122 *Packet Telephony Farm Card – High Volume*) card processes TDM samples from VMU. It works as a *Media Gateway*. It contains: DSP MSC 8103 (aggregator), FLASH memory, 5 × (DSP MSC8122 + SDRAM memory), etc.

Basic VMU board functions are: generating of voice messages in 120 channels, connection with IMG board over RS232 or FE, messages content change over FE, etc.

IMG95 is packed in a 19" x 1 U x 200 mm rackable chassis, with 230 V AC power supply and 20W consumption.

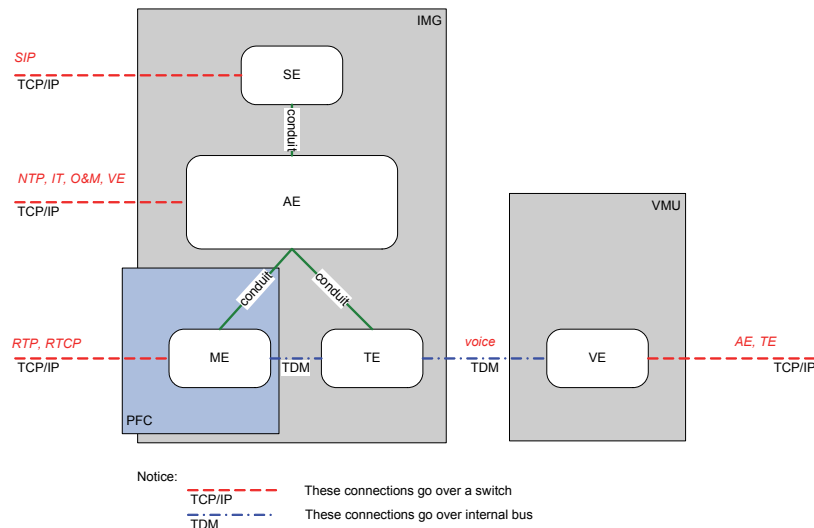


Fig. 2. Scheme of IMG95 structure.

V. ETS SOFTWARE

IMG95 uses *embedded* Linux – modified *Freescale* version. Linux and development environment used are:

- Uboot 2.0,
- kernel 2.4.20.

Application IMG software consists of engines, which are part of applied software packages [4].

VeriCall administrator is a part of complex software package which maintains the administrative engine – AE, media engine – ME and connects all engines by conduit – TCP/IP mechanism for data exchange.

SE is a software package containing:

- SIP user agent UA,
- SDP module – support of configuration of SIP subscribers defined by SDP protocol [5],
- SIP client and SIP server,
- SIP outgoing and incoming multiplexer.

TE is an application performing functions:

- TDM channel control: signaling with VMU,
- VMU control: communication with VMU.

ME is a part of *VeriCall* software package dealing with media engine. It has functions:

- *VeriCall* functions: codec's and VQE functions
- Communication with AE,
- Communication with TDM channels,
- RTP/RTCP traffic over TCP/IP,
- Aggregate functions: initialization and booting of DSP's, traffic control over packet I/O.

VMU (*Voice Messages Unit*) is a voice machine with functions:

- Generation of voice messages in 120 channels,
- Communication with TE,
- Synchronization with *ntp* service,
- Loading of voice messages elements over TCP/IP.

O&M functions on computer NU consist of:

- statistical functions,
- alarm functions,
- archive functions,

- help functions,
- web tools.

Each of the above functions has its own window for interaction with operator. Additionally, NU statistics and alarms are transferred on-line to a database on another server, and can be accessed using the web interface from any computer on IT network. This part of functional software was made by Telekom Serbia engineers.

VI. EXPLOITATION EXPERIENCES

After passing internal Telekom Serbia testing, and gaining a Technical Permit – Certificate on compliance with prescribed standards and regulations from RATEL (Republic Agency for Electronic Communications), ETS was put into operation in NGN of Telekom Serbia.

For IRITEL the most significant result was the fact that with ETS the VoIP technology was mastered with good results. Based on IMG, a family of VoIP devices is conceived. Some of member of this family can be:

- IAM-MSAN – *Multiservice Access Node* – up to 4000 subscribers,
- IAD – *Integrated Access Device* – up to 32 subscribers,
- IPPBX – up to 500 subscribers,
- DKTS-AN – *turning TDM digital switch DKTS into an access node* - up to 4000 subscribers,
- IVR – *Integrated Voice Response* – up to 600 subscribers,
- Call Center – up to 600 operators.

IMG family will be a basis for a massive production of NGN access devices.

Exploitation showed some benefits over the previous solution:

- elimination of costly PRI trunks, as the system is connected with NGN over IP by SIP trunking,
- system is entirely in NGN, no connections with PSTN/PLMN (no unnecessary voice conversions),
- synchronization with two redundant *ntp* servers ,
- increased reliability,
- system is more resilient to “bursty” traffic loads,
- system is cheaper (being a domestic solution).

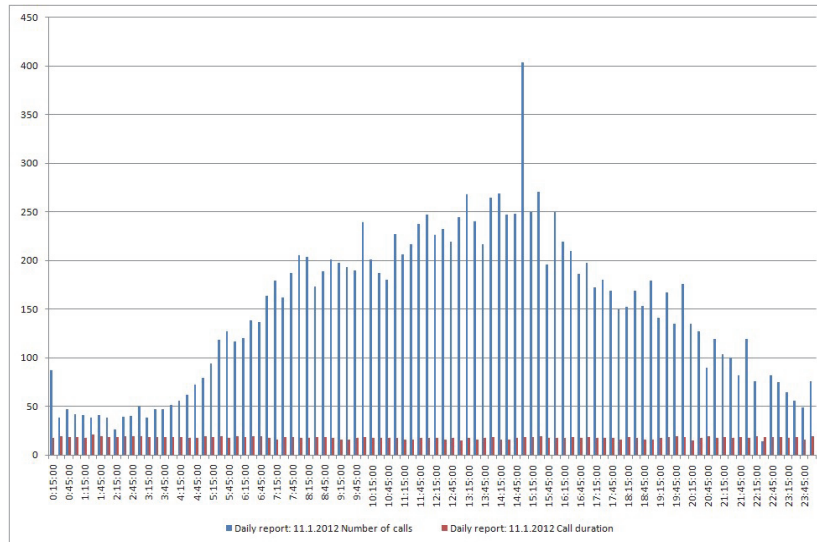


Fig. 3. Call dispersion for 11.01.2012.

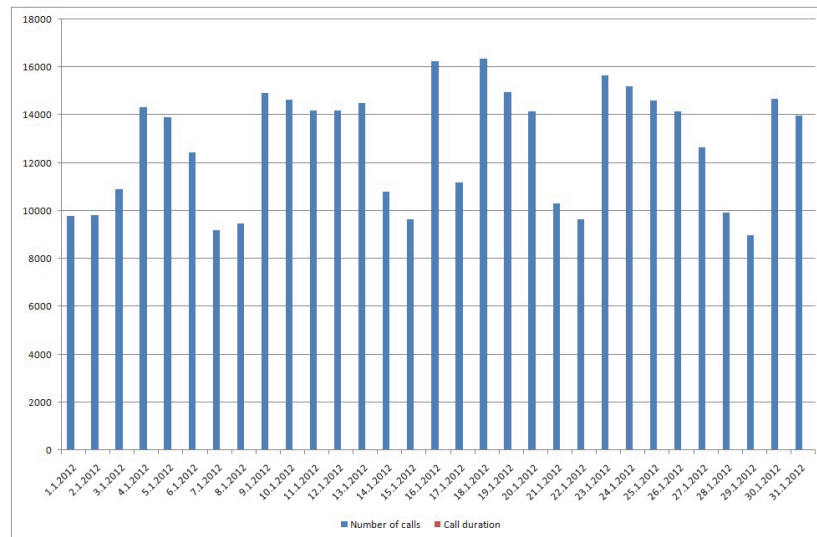


Fig. 4. Call statistics for January 2012.

The following figures show the behavior of ETS in exploitation. Fig. 3 shows the statistical dispersion of calls during the day 11.1.2012, and Fig. 4 shows call statistics for January 2012.

We can see that the number of calls per day is 14.000 on the average, and around 400.000 per month. An average call duration is 16 s. The system is most busy between 15:00 and 15:15 hours, and BHCA (*Busy Hour Call Attempts*) is 1173 calls from 14:45 to 15:45 h. The number of busy channels during BHCA is 30. The most busy day is Monday, and the ratio: working day/weekend day is approximately 1.57.

Fig. 5 shows the number of calls and busy channels during New Year's Eve, a typical "bursty" situation. We can see that the system behaves well under extreme traffic load – BHCA was from 23:30 to 00:30 with 2862 calls over 96 busy channels.

Another "heavy load" day was 25.3.2012, when the *Daylight Saving Time (DST)* change of time occurs. The number of calls was 65257, with BHCA from 8:30 to 9:30 with 6631 calls.

VII. TRAFFIC POSSIBILITIES OF A GROUP OF 120 CHANNELS

In technical requirements for the system it was defined that it should have 120 channels which must accept 80 Erl of telephone traffic. Let us assume that the possibility of occupancy of all 120 channels is not greater than 0,00001, that is that 1 call out of 100000 is not able to be serviced. This level of service is far better than a level defined for telephone exchanges [6], and it approaches the most severe conditions defined for CCS7 network [7]. For the largest allowed incoming traffic of 80 Erl we have blocking probability:

$$B(120,80) = ERL(120,80,120) \approx 0,00001$$

where:

$$ERL(j, A, N) = (A^j / j!) / \left(\sum_{k=0}^N A^k / k! \right)$$

is the Erlang's possibility of occupying j of N channels if offered traffic is A .

So, if the greatest offered traffic is 80 Erl, we can find the allowed time of channel occupancy (t_m) and calling intensity (λ) from the formulae:

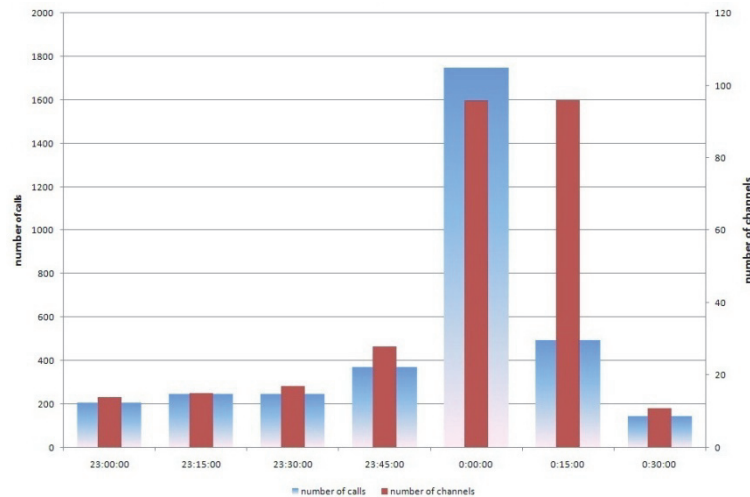


Fig. 5. Call statistics for New Year's Eve.

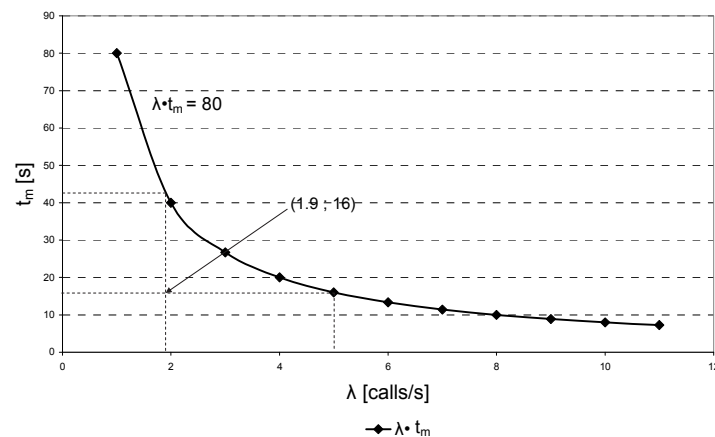


Fig. 6. Traffic possibilities of a group of 120 channels.

$$\lambda = A/t_m$$

and

$$\lambda t_m \leq 80.$$

This border function is shown in Fig. 6. The greatest measured calling intensity - the one in the New Year's Eve ($\lambda=1,9$ calls per second) and the mean time of channel occupancy ($t_m=16$ seconds) define a point which is far from the border of allowed load – Fig. 6.

If the mean time of occupancy is 16 s, the greatest offered number of calls per second could reach $\lambda = 5$.

For the greatest measured calling intensity $\lambda=1.9$ calls/s the mean time of occupancy could reach $t_m = 42$ s.

We can see that the system is well designed for burst traffic, and can withstand heavy load without performance degrading. It can fulfill the present traffic needs, and significant increases in traffic in the future.

VIII. CONCLUSION

The paper shows that ETS is well designed and built in such a way that it can easily fulfill the present and future traffic demands. The significance of ETS lies in the fact that it is one of the rare completely domestic solutions in VoIP technology domain. It is one of the family of IMG devices based on media gateway enabling a breakthrough in the IP world. IRITEL is hoping to offer a new, cheaper, domestic product which could replace imported solutions. One of the important implementations of this platform

would be a DKTS-AN, which could revitalize domestic switching system DKTS turning it into an access node for NGN and introducing new features for its subscribers [8]. It is a step towards a new telephony architecture [9].

ACKNOWLEDGMENT

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