Cloud Technologies Application at English Language Studying for Maritime Branch Specialists

Vladlen Shapo [1[0000-0002-3921-4159]], Valeriy Volovshchykov [2[0000-0003-4454-2314]]

[1] National University "Odessa Maritime Academy", Didrihsona str. 8, 65029 Odessa, Ukraine
[2] National Technical University "Kharkiv Polytechnic Institute",
Kyrpychova str. 2, 61002 Kharkiv, Ukraine
[1] vladlen.shapo@gmail.com
[2] valera@kpi.kharkov.ua

Abstract. Last decade true jump in complex technical systems developing is occurred. In maritime field technologies based on Industry 4.0, IoT, IIoT concepts are implementing actively. Thanks to progress in wireless and satellite technologies, mass embedded computers and PLCs appearance it's possible to solve totally new tasks like fully unmanned ships creation. A lot of seafarers work in the sea and communicate with colleagues mostly in English language and have to improve its level nonstop. Without satisfactory level of English language knowledge it's impossible to get any job in maritime branch. Modern development directions also require improving of IT terms knowledge. By IMO statistics near 80% of accidents in the sea happen because of human errors. Some of them are caused by misunderstanding between people, including language problems. In this paper it's based the cloud platform creation which will complement possibilities of existing platforms and will allow seafarers to communicate directly, making their language studying more diversified. Proposed approach is more flexible in terms of cost, number and activity of participants and, respectively, hardware and Internet channels workload. Very important aspect of proposed approach is multimedia traffic transferring providing. Mathematical expressions for calculations automation and systems and channels characteristics formalization choosing are proposed as well.

Keywords: Cloud Technologies, Industry 4.0, IoT, IIoT, Sound And Video Traffic, English Language Studying, Seafarers.

1 Aim

Last decade take place true jump in approaches to developing, control and exploitation of different complex technical systems. In maritime field data exchange technologies, based on Industry 4.0 (4th industrial revolution), IIoT (Industrial Internet of Things), Shipping 4.0 (revolutionary new approaches to control of the ships, their development and exploitation), etc. concepts are implementing very actively. Thanks to vast progress in wireless and satellite data transfer technologies and mass appearance of high performance intellectual embedded computer systems and information technologies it's became possible to maintain inextricable links to the classical approaches, continuing exploitation of existing systems with upgrading them step by step or fully replacing to more modern ones, and at the same time to create most complex technical systems like even fully unmanned ships, and also allow to perform absolutely new tasks like truly intelligent remote control of fully unmanned ships and complex technical systems.

Thereby Unmanned Cargo Ship Development Alliance [1] is founded; Advanced Autonomous Waterborne Applications Initiative [2] autonomous ship research project and Maritime Autonomous Surface Ships [3] direction are created; Distributed Intelligent Vessel Components (DIVEC) software, providing new protocol for devices connecting and data transferring is developed [4]; Digital, Internet, Materials & Engineering Co-Creation (DIMECC) technical ecosystem [5] and One Sea Ecosystem Alliance [6] are created. Members of these alliances and initiatives are big innovative branch leading companies. As a result of these efforts on maritime branch new digital transformation plans are appeared, which envisage creation of unmanned, autonomous and remote controlled in various degrees ships by 2025 - 2035.

In maritime branch tens and even hundreds of thousands professional seafarers, thousands of full time students, students by correspondence, postgraduate students, advanced training courses students and trainees every year work in the sea on board a ships or pass many month naval training on different type of commercial vessels and need to communicate with another crew members from different countries mostly in English language and have to improve its level being far from the home and often without any teacher’s consultations. Without satisfactory level of English language knowledge it's impossible to get any job in international maritime shipping or crewing companies, in maritime logistics sector, on board a ship, offshore or onshore subdivisions. Mentioned above development directions [1-6] also require significant enhancement of English language knowledge.

There are some functioning English language studying platforms for seafarers: Marlins Test Platform, SeaTALK, MarTEL - Maritime Tests of English Language and MarTEL Plus, Captains Maritime Training [7-10]. Mentioned English language studying platforms for seafarers propose different training solutions and possibilities with games, animations, real life situations, virtual environments, etc. At present time there are no international standards for evaluation of the English language skills for seafarers. International Maritime Organization (IMO) statistics says that about 80% of accidents in the sea happen because of human errors. Some of these accidents or critical situations may be caused by misunderstanding between people, including language problems.

Content from different platforms is available partly in online, partly in offline (after downloading without Internet connection necessity) modes. But main lack is that these resources don't allow to communicate seafarers from different countries and with different specialties directly within the same platform in real time to refresh some skills, to taste different accents and pronunciations and speaking style, far from ideal. Also these platforms are quite static, created for standard studying conditions and classic fields (mostly navigation) and can't be quickly adjusted to dynamically changed environment (newly created technical development directions and, respectively, new terms, concepts and abbreviations).

In this paper it's based the cloud platform creation which will complement possibilities of existing platforms and will allow seafarers to communicate directly, making their trainings much more diversified. Also proposed approach is more flexible in terms of cost, number and activity of participants (and, respectively, hardware and Internet channels workload) and will allow participants to create own resources. Very important aspect of proposed approach is multimedia traffic transferring providing.

2 Findings

There are two main ways to organize remote access to any cloud system. First way means that all concerns will take over provider of cloud service (most known are Microsoft Azure, Amazon Web Services, Google Cloud Platform). But all proposed possibilities may be paid by customer. Also all data are placed physically outside of corporate computer network (CCN).

Second way means that all hardware are installed inside of CCN. All data are placed physically inside CCN, but organization has to ensure a lot of different requirements by its own strength.

Both ways have positive and negative sides, advantages and disadvantages at analyzing from different points of view. All possible parameters are presented in Table 1.

Table 1. Comparison of two approaches on information system building.

#	Requirement	Cloud	Inside of CCN
1	Presence of own hardware	Not necessary	Necessary
2	Internet channels	CSP	Necessary
3	Uninterruptible power supply	CSP	Necessary
4	Physical protection of hardware	CSP	Necessary
5	Electricity providing	CSP	Necessary
6	Software installation	CSP	Necessary
7	Software upgrade	CSP	Necessary
8	Cyber security providing	CSP	Necessary
9	Highly qualified system administrator position searching	CSP	Necessary
10	Highly qualified system administrator expenses	CSP	Necessary
11	IT team positions searching	CSP	Necessary
12	IT team positions expenses	CSP	Necessary
13	First stage expenses	Depends on situation (big/rational/moderate)	Big
14	Maintenance expenses	Depends on situation (big/rational/moderate)	Depends on situation (big/rational/moderate)
15	Backups/recovery	CSP	Necessary
16	Data placement	Cloud	CCN

Final choice depends on concrete situation and on variety of additional factors but in this paper it's recommended to use completely cloud approach because only in this case proposed information system may be absolutely independent from any state or private organization excluding cloud services provider (CSP) and may be much more flexible and dynamically changing in accordance with surrounding conditions and external requirements.

Fig. 1 presents cloud approach at English language studying information system developing (physical hardware is hosted by CSP or all requirements are supplied by CSP, data are placed in the cloud).

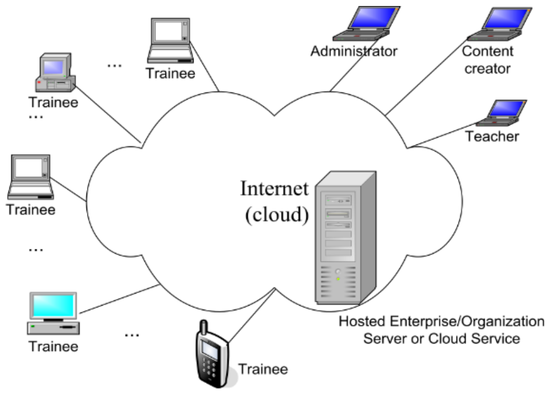


Fig. 1. Cloud approach at English language studying information system developing.

Fig. 2 presents second way at English language studying information system developing (physical hardware is placed inside CCN, data are placed in CCN).

Own experience during nine years period on Learning Management System maintenance [11] shows that cloud approach has nowadays much more advantages. That's why in this paper it's proposed to use exactly cloud approach.

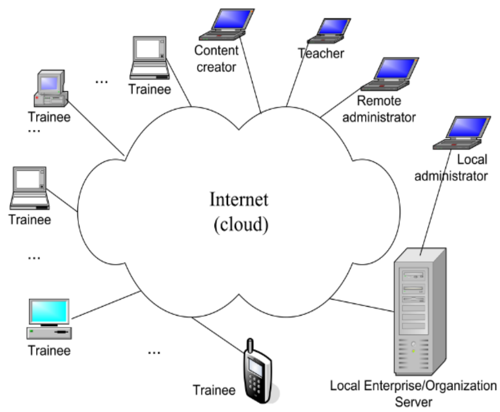


Fig. 2. Placement of hardware inside the CCN at English language studying information system developing.

At design, installation, exploitation and modernization of any modern data transfer network, which has to transfer sound/voice/video traffic, it's necessary to take into consideration variety of different standards and protocols, which are created by different proprietary developers and with participation of ISO, IEEE ITU and another standardization organizations. Variety of protocols supporting will allow to attract a lot of users which have absolutely incompatible or obsolete hardware and software (operating systems and application software).

At present time there are four groups of VoIP (Voice over IP) protocol families: SIP, H248, H323, SIGTRAN. These groups are briefly analyzed below.

SIP protocols family based on following IETF (Internet Engineering Task Force) specifications. SIP (Session Initiation Protocol) used for call control (the call control protocols establish, modify and release connections). It is an application-layer control (signaling) protocol for sessions. These sessions include Internet telephone calls, multimedia distribution, and multimedia conferences. SIP can create, modify, and terminate sessions with one or more participants. SDP (Session Description Protocol) used to describe the "media" session inside of SIP and some other protocols. RTSP (Real Time Streaming Protocol) used to set up streaming sessions. SigComp (Signaling Compression) is a solution for compressing messages generated by application protocols such as SIP and RTSP.

H.248 group is based on following IETF and ITU-T specifications. H248/MEGACO: "MEdia GAteway COntrol protocol" H248.1 version 3. RFC (Request for Comments) 5125. H.248.1 Version 3 ITU-T H.248. MGCP – Media Gateway Control Protocol RFC3435.

H.323 group is based on following ITU-T (International Telecommunication Union, Telecommunication standardization sector) specifications. H223 – multiplexing protocol for circuit-based multimedia communications system. H225 – call control protocol. H235 – security. H245 – media control protocol. H323 – packet-based multimedia communications systems. H450 – supplementary services. Q.931 – ISDN (Integrated Service Digital Network) user-network interface layer 3 specification for basic call control.

SIGTRAN group is often used in connections towards PSTN (Public Switched Telephone Network).

Most popular proprietary protocols are briefly described below.

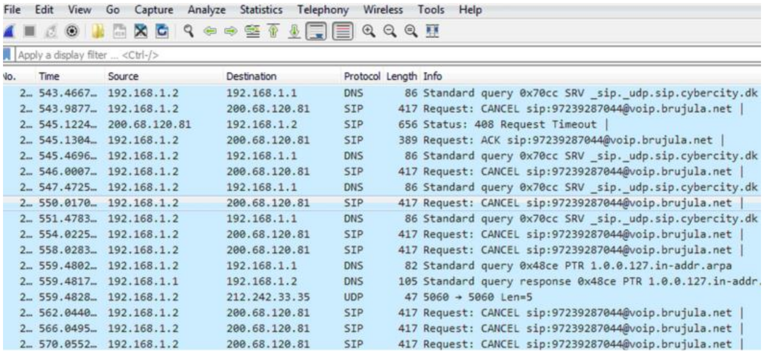
Cisco SKINNY is terminal control protocol.

Digium IAX2 (Inter-Asterisk eXchange version 2) is standalone VoIP protocol primarily developed for communication between Asterisk servers.

Nortel UNISTIM (Unified Networks IP Stimulus) is stimulus protocol for IP phones.

Using of wide spectrum of different devices, operating systems, application software, network equipment, etc. potentially may cause problems with compatibility and network faults. In these situations it's recommended to use corresponding professional software to analyze network traffic (for example, Wireshark). Sample of some protocols traffic captures is shown at Fig. 3.

Some possibilities, subtleties and application of Wireshark software are described in papers [12].



No.	Time	Source	Destination	Protocol	Length	Info
2…	543.4667…	192.168.1.2	192.168.1.1	DNS	86	Standard query 0x70cc SRV _sip._udp.sip.cybercity.dk
2…	543.9877…	192.168.1.2	200.68.120.81	SIP	417	Request: CANCEL sip:97239287044@voip.brujula.net |
2…	545.1224…	200.68.120.81	192.168.1.2	SIP	656	Status: 408 Request Timeout |
2…	545.1304…	192.168.1.2	200.68.120.81	SIP	389	Request: ACK sip:97239287044@voip.brujula.net |
2…	545.4696…	192.168.1.2	192.168.1.1	DNS	86	Standard query 0x70cc SRV _sip._udp.sip.cybercity.dk
2…	546.0007…	192.168.1.2	200.68.120.81	SIP	417	Request: CANCEL sip:97239287044@voip.brujula.net |
2…	547.4725…	192.168.1.2	192.168.1.1	DNS	86	Standard query 0x70cc SRV _sip._udp.sip.cybercity.dk
2…	550.0170…	192.168.1.2	200.68.120.81	SIP	417	Request: CANCEL sip:97239287044@voip.brujula.net |
2…	551.4783…	192.168.1.2	192.168.1.1	DNS	86	Standard query 0x70cc SRV _sip._udp.sip.cybercity.dk
2…	554.0225…	192.168.1.2	200.68.120.81	SIP	417	Request: CANCEL sip:97239287044@voip.brujula.net |
2…	558.0283…	192.168.1.2	200.68.120.81	SIP	417	Request: CANCEL sip:97239287044@voip.brujula.net |
2…	559.4802…	192.168.1.2	192.168.1.1	DNS	82	Standard query 0x48ce PTR 1.0.0.127.in-addr.arpa
2…	559.4817…	192.168.1.1	192.168.1.2	DNS	105	Standard query response 0x48ce PTR 1.0.0.127.in-addr.
2…	559.4828…	192.168.1.2	212.242.33.35	UDP	47	5060 → 5060 Len=5
2…	562.0440…	192.168.1.2	200.68.120.81	SIP	417	Request: CANCEL sip:97239287044@voip.brujula.net |
2…	566.0495…	192.168.1.2	200.68.120.81	SIP	417	Request: CANCEL sip:97239287044@voip.brujula.net |
2…	570.0552…	192.168.1.2	200.68.120.81	SIP	417	Request: CANCEL sip:97239287044@voip.brujula.net |

Fig. 3. Sample of SIP protocol traffic capture in Wireshark software.

Fig. 4 presents model of voice/sound traffic exchange between individual users and users workgroups using cloud technologies $(V_{t1}, V_{t2}, ..., V_{ti}, ..., V_{tn-1}, V_{tn})$ is vector of different voice/sound protocols data volumes.

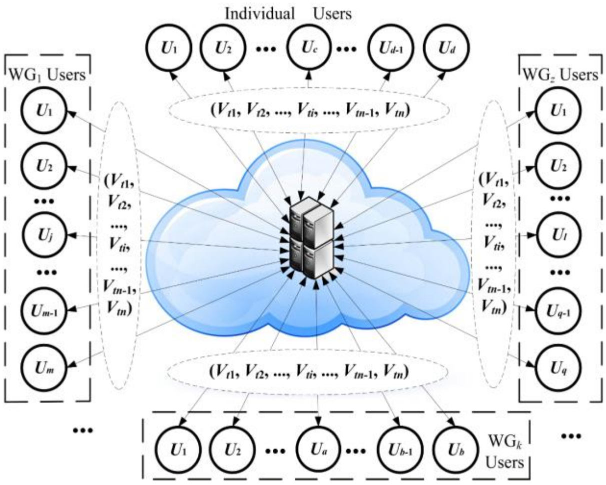


Fig. 4. Model of voice/sound traffic exchange between individual users and users workgroups using cloud technologies.

When seafarers work in a sea they may use only satellite Internet connection. One of most popular satellite data transfer technologies is VSAT (Very Small Aperture Terminal), which may be used for two direction communications. Data transfer rates are available in the typical range from 64 kbps up to 8 Mbps, but slower and faster data transfer rates are possible as well.

Traditional maritime communication systems require vessels to share bandwidth. But this way is inefficient at transferring the huge data amounts which is actual nowadays. That's why VSAT can guarantee bandwidth level for data or voice transmission by using of specially reserved channels.

Delays at long distance voice transfer affect the speech quality in high degree. Main latency values and transferred voice quality are compared in Table 2.

Table 2. Latencies influence at voice signal perception.

Latency, ms	Voice connection quality
> 600	Interaction is impossible
600	Interaction is difficult
250	Voice distortion level is acceptable
100	Delays are almost discreet
50	Voice data transmission with no distortion

Also it's necessary to take into consideration concrete protocols and their characteristics like header and trailer values, payloads and so on. Some most popular protocols are analyzed below.

1. IP header: 40 bytes.
2. User Datagram Protocol header: 8 bytes.
3. Real-Time Transport Protocol header: 12 bytes.
4. Multilink Point-to-Point Protocol (MP) or Frame Relay Forum .12 (FRF.12) Layer 2 header: 6 bytes.
5. End-of-frame flag on MP and Frame Relay frames: 1 byte.
6. Ethernet L2 headers: 18 bytes (including 4 bytes of Frame Check Sequence or Cyclic Redundancy Check).

At the same time Compressed Real-Time Protocol (cRTP) reduces the headers of mentioned above protocols (1-3) to 2 or 4 bytes (it is not available over Ethernet).

Thanks to fast growing of processors productivity and efficient algorithms development new and effective codecs became available nowadays. Codecs are used to convert voice signal from analogue to digital form and significantly vary in result sound quality and the bandwidth required. Different programs and devices may support several different codecs and have to negotiate which one will be used. Mentioned in Table 3 codecs may be implemented at voice transmitting.

Table 3. Most popular codecs and their characteristics.

	Characteristics of the codec		
Codec name and bit rate, kbps	Sample size, Bytes	Sample interval, ms	Mean opinion score
G.711 (64)	80	10	4.1
G.729 (8)	10	10	3.9
G.723.1 (6.3)	24	30	3.9
G.723.1 (5.3)	20	30	3.8
G.726 (32)	20	5	3.8
G.726 (24)	15	5	NA
G.728 (16)	10	5	3.61
G722_64k (64)	80	10	4.13
ilbc_mode_20 (15.2)	38	20	NA
ilbc_mode_30 (13.33)	50	30	NA

Table 4. Most popular codecs and necessary bandwidths

Codec name and bit rate, kbps	Bandwidth values					
	Voice payload size, Bytes	Voice payload size, ms	Packets per second	Band-width MP or FRF.12, kbps	Band-width w/cRTP MP or FRF.12, kbps	Ethernet Band-width, kbps
G.711 (64)	160	20	50	82.8	67.6	87.2
G.729 (8)	20	20	50	26.8	11.6	31.2
G.723.1 (6.3)	24	30	33.3	18.9	8.8	21.9
G.723.1 (5.3)	20	30	33.3	17.9	7.7	20.8
G.726 (32)	80	20	50	50.8	35.6	55.2
G.726 (24)		20	50	42.8	27.6	47.2
G.728 (16)	60	30	33.3	28.5	18.4	31.5
G722_64k (64)	160	20	50	82.8	67.6	87.2
ilbc_mode_20 (15.2)	38	20	50	34.0	18.8	38.4
ilbc_mode_30 (13.33)	50	30	33.3	26.86	15.7	28.8

Table 3 and Table 4 contain some specific terms, described below.

1. Codec bit rate. It is the number of bits per second, which have to be transmitted to deliver a voice call. May be obtained as result of division of codec sample size by codec sample interval. Unit of measurement is kbps.
2. Codec sample size. It is the number of bytes, captured by the digital signal processor at each codec sample interval. For example, the G.729 codec operates on sample intervals of 10 ms, which corresponds to 10 bytes (80 bits) per sample at a bit rate of 8 kbps. Unit of measurement is Byte.
3. Codec sample interval. It is the sample interval at which the codec operates. Unit of measurement is ms.
4. Mean opinion score (MOS). This parameter used to grade the voice quality of telephone connections. Using this parameter listeners judge the quality of a voice sample on a scale of one (bad) to five (excellent). The scores are averaged in order to provide the MOS for the codec.
5. Voice payload size. This characteristic represents the number of bytes or bits that are filled into a packet. The voice payload size must be a multiple of the codec sample size. For example, G.729 packets can use 10, 20, 30, 40, 50 or 60 bytes of voice payload size. Unit of measurement is Byte or bit.
6. Voice payload size can also be represented in terms of the codec samples. For example, a G.729 voice payload size of 20 ms (two 10 ms codec samples) represents a voice payload of 20 bytes: 20 bytes × 8 / 20 ms. Result is 8 kbps. Unit of measurement is ms.
7. Number of packets per second (pps). This value represents the number of packets, which have to be transmitted every second to provide the codec bit rate. For example, for a G.729 call with voice payload size per packet of 20 bytes (160 bits),

50 packets need to be transmitted every second: 8 kbps / 160 bits per packet. Result is 50 pps.

Right combination of protocols and codecs implementation allows to enhance quality of interaction, to reduce data transmitting volume, price/productivity ratio and general expenses.

Let in data transfer network (local or based on cloud technologies) it's necessary to process voice/sound data, incoming from n devices, appliances and so on. In the general case these devices it's possible to subdivide to t subtypes for getting of exact values of each subtype devices influence for network and equipment with digital interfaces loading during network exploitation or its upgrading planning, when certain number (k) of each subtype devices exists already or may be planned for installation, meanwhile $n = \sum_{i=1}^{t} k_i n_i$. In the general case volume of transferring data V_f and minimal demanded data transfer network segment or internetwork (Internet) channel bandwidth B_f are accordingly

$$V_f = \sum_{i=1}^{n} V_i \quad (1)$$

$$B_f = \sum_{i=1}^{n} B_i \quad (2)$$

where V_i – volume of data, transferring to control computational system (local or cloud) from i-number device.

B_i – data transfer network bandwidth, demanded for data transferring from i-number device.

Situation is possible, when some same type devices transfer identical data volumes to local or cloud control computational system and create identical network segment or channel loading. In this case volume of transferring data V_f and minimal demanding network segment or internetwork (Internet) channel bandwidth B_f it's possible to count using following formulas

$$V_f = \sum_{i=1}^{n} k_i V_i \quad (3)$$

$$B_f = \sum_{i=1}^{n} k_i B_i \quad (4)$$

where k_i – number of same type devices of i-type, which generate identical volume of data, transferring to local or cloud control computational system, creating similar network loading.

Let in the network structure there are m control computational system (local), which transfer data to central control computational system (cloud). In this case volume of data V_{fc}, transferring to central control computational system and minimal demanding bandwidth B_{fc} of corresponding network segment or internetwork (Internet) channel are accordingly

$$V_{fc} = \sum_{k=1}^{m} V_{fk} \quad (5)$$

$$B_{fc} = \sum_{\mathrm{k}=1}^{m} B_{fk} \quad (6)$$

where V_{fk} – volume of data, transferring to cloud central control computational system from local control computational system number k;

B_{fk} – network bandwidth, demanding for data transfer from control computational system number k.

Implementation of one or more local control computational system allows to reduce expenses for cloud system model choosing taking into consideration following parameters: necessary processor(s) productivity, random access memory volume, data store volume and productivity and to minimize expenses for Internet channel rent. Local control computational system may be used if some users use the same network (cloud) service but simultaneously work in the same local network.

Formulas (1)-(6) allow to calculate minimal necessary bandwidths and overall loading of Internet or CCN channels and sound and video data volumes which have to be transferred between different users.

3 Conclusion

Complex approach devoted to development of universal language studying cloud platform is described in this paper. Proposed formulas may be used as the base of mathematical model at calculating of volumes of data, which have to be transferred by the Internet or CCN channels, minimal necessary bandwidths and overall loading of network channels and network equipment. It allows to automate optimal choice of Internet or CCN standard bandwidth at their rent or creation and also to automate productivity of network equipment and cyber security systems hardware taking into account expenses/productivity ratio. Also analysis of modern protocols for sound and video transferring is made and recommendations on bottlenecks searching using corresponding software at platform development and maintenance are offered.

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