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SCOPE EXTENSION IN A CONVERGED NETWORK

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Background. Unified IP-based provisioning of communication services often faces the problem of QoS guarantees during the real-time application traffic transmission. The packet loss and random time delays in statistically multiplexed data flow apparently make a lot of trouble during remote audio/video conversation or machine-to-machine interaction in real time mode. Known IP-based approaches in telecom network engineering can't meet the highly increasing operator demands and consumer requirements to QoS-aware services; this hinders the Internet and telephone networks convergence with respect to quality of service provision. Therefore, more research needed to effectively address these problems.

Objective. The aim of the paper is the interoperability scope extension in a converged telecom network to provide real-time data transfer with quality of service support.

Methods. The current state of "Internet+telephone network" along with the synthesis of an advanced interface for a heterogeneous network is analyzed.

Results. Basic principles introduced to extend the autonomous network interoperability. A novel mechanism of conveyor transporting modules originated for a converged network to benefit packet and circuit switching unification.

Conclusions. Traditional telecoms and Internet service providers permanently compete for IT market niches having their pros and cons, and vast IP-applications merged on the market. Therefore, no urgent successor-protocol reasonable in a short term. On this agenda, an alternative algorithm proposed for dynamic packet data allocation in a unified network-to-network link, which extends the interoperability scope in a converged network. The Gigabit Ethernet linkage outlined based on proposed algorithm.

Keywords: networks interoperability; QoS; real-time applications.

Introduction

The impressive achievements in communication networks convergence have been widely exhibited in last decades with particular efforts on telephony and Internet service integration The ITU-concept of next generation network (NGN) architecture implies unified service provision on IP Multimedia Subsystem platform (IMS), whereas packed based transport evolving to generalized MPLS (GMPLS), along with internetworking enhancement via conventional IPv6 protocol deployment, [1-2]. This strategy of telecom network advancement seems to be an acceptable and balanced compromise between arising quality of service requirements and existing communication infrastructure.

The known solutions in telecom systems and networks are mainly focused on maximum utilization of ubiquitously commercialized technologies and standard protocols. However, consistent development of a truly unified transporting platform for a wide area heterogeneous telecom network is challenging yet. The specialists eventually come to understand the inevitable constrains of generic internetworking mechanisms and related protocols of TCP/IP suit [3]. In fact, the open system interconnection model based on internet protocol IPv4 or IPv6, was developed a half century ago, and hence, not primarily aimed for real-time traffic support. This shortcoming inhibits the seamless integration of diverse autonomous systems; in particular, this limits to merge conventional public switched telephone networks (PSTN) and Internet. On the other hand, a vast multimedia application inundates today's IT market, implicating the usage of underlying IP-packing layer.

Known IP-based approaches in telecom network engineering seems can't meet the highly increasing operator demands and consumer requirements to the autonomous systems interoperability, with respect to the future IT services, including the fully packet based digital telephony and QoS sufficient multimedia, fast dynamic machine-to-machine cooperation and extremely growing Internet of things. Therefore, new comprehensive researches needed to substantiate a holistic long term vision on network system architecture, incorporating the most fundamental theoretic contributions and high-tech advancements in telecom realm.

Statement of objective

Considering aforesaid reasoning, this paper targets the interoperability scope extension in converged telecom network to provide real-time data transfer with quality of service support.

The issues of telephony and packet based network convergence

The core issue of PSTN-to-Internet framework convergence is generic counteracting among two controversial techniques of data multiplexing, which commonly referred to as Time Division Multiplexing (TDM) and Statistical Multiplexing (SMP). The TDM digital channels are mostly used for real-time traffic delivery with guaranteed quality of service provision (i.e. the standard telephony). The long haul regional and global telecoms also provide conventional Internet data streaming over the aggregated TDM trunks. Instead, the SMP data transfer has been decisive immanent factor of packet based networking paradigm originated by L. Kleinrock and P. Baran, who anticipated the modern Internet emerge.

The TDM telecoms and STM companies permanently compete for IT market niches having their own evident pros and cons in particular services. The TDM benefits the QoS ensured telephone connection on guaranteed bandwidth and constant bit rate (CBR) via the typical 64 Kbit/s linkage of the E0 channel. Though, the TDM suffers an exclusive channel reservation with no care about the competence of given capacity utilization. In contrast to TDM multiplexing the SMP provides comprehensive utilization of channel resource approaching the 90%–100% efficiency in maximum load. This eventually resulted in critically low cost traffic billing as a main privilege of Internet aided communication. In turn, chaotic packet loss and random time delays in statistically multiplexed STM linkage apparently make a lot of trouble while information exchange in real time mode.

The current digital telephony standard requires the one way delay metric (OWD) for voice over TDM circuit within 100 ms [4]. Conventional TDM communications widely exploit generic SONET/SDH (Synchronous Optical Network / Synchronous Digital Hierarchy) technology with either T0 (American) or E0 (Europe) digital signal bearers intended for distinct audio communication linkages.

The T0(E0) bearer carries the 7(8)-bit voice units with 8 KHz sampling. The voice quantization time in T0/E0 data flow is fixed by 1/8KHz=0.125ms; this determines maximum voice sampling latency of about 0.125 ms while telephone session over the T0/E0 remote connection. It is clear, that 0.125 ms of one way voice latency, as well as resulting 2×0.125=0.25 ms of two way delay (TWD), will merely affect the overall conversation performance compare to 100 ms OWD or 200 ms round trip time delay.

In contrast to regular voice data transmission over the TDM end-to-end connection, the STM based voice communication uses IP-packet bearers encapsulated in user datagram transporting protocol units (UDP). Hereby, streaming voice data arranged in the sequence of audio sample frames (ASF), which online processed by voice codec (vocoder) that analyzes and synthesizes the human voice signal for audio data compression, multiplexing and encryption,. The vocoder output product is a sequence of encrypted and compressed voice data segments (VDS).

The information capacity and time duration of ASF depend on vocoder compression ratio k_{VC}. For instance, a typical G729 vocoder shows k_{VC}=8. Thus, any compressed VDS of 20 octets corresponds backwards to original 160-octet audio sample frame ASF; hereby, ASF duration is (0.125 ×160)ms=20ms. The one-way delay of such a data unit is approximately estimated by the following empirically obtained formula: OWD = $(20+\Delta t_{Co}+\Delta t_{Tr}+20+\Delta t_{DCo})$ ms. Herein two adds of 20 ms mean audio sampling time (speaker party) and playing back time (listening party); the two adds Δt_{Co} and Δt_{DCo} mean ASF encryption/decryption time, which depend on the ASF size and vocoder performance; the Δt_{Tr} means VDS delivery time while end-to-end VDS streaming over the packet based network. Let Δt_{Co} and Δt_{DCo} take about 5ms each, then one-way delay resulted in OWD≈$(50+\Delta t_{Tr})$. This formula shows the critical importance of voice data delay Δt_{Tr} as a principal component of the overall OWD metric while being transported over the packet based network.

Concerning the QoS requirements in terms of one-way delay, the aforesaid component of transporting voice latency Δt_{Tr} cannot be actually limited by 50 ms option to satisfy the overall 100 ms OWD. Indeed, even omitting packet switching delays, the solely optic wave propagation may need about 75 ms in a global network connection, [5]. The specialists suppose 150 ms OWD could be an acceptable compromise for the future global packet based telephony of high quality, [3-4], [6].

The overall latency of real-time data delivering through a packet based telecommunication network can be composed of two decisive parts. The first one includes consolidated forwarding time summarizing all the internal paths of those autonomous systems which provide given end-to-end virtual circuit. The second part of the overall transferring data latency appears between adjacent autonomous systems at the interworking edge links. Presently, various mechanisms developed for high speed packet delivery within a local domain of an autonomous system, mostly based on GMPLS aware switches or similar proprietary networking equipment [2] that operate on the intermediate level between the data link and network layers (e.g. layers L2.5 and L3 in terms of conventional OSI/ISO model).

Nevertheless, the autonomous system interoperability still remains in the competence of network layer routing devices driven by the unique and commonly adopted border gateway protocol BGP-4, [7]. Actually, these very segments of any long haul inter domain transportation path traversing the Internet cloud, consider the most critical bottlenecks, which cause QoS troubles while IP packet delivery. The status quo of contemporary Internet design seriously hampers the long-term perspective for heterogeneous network convergence with respect to TDM based public switched telephone networks and STM based packet-switched IP networks. Another to say, the current IP paradigm as the core protocol of entire TCP/IP stack, however, turns to be the most conservative component of Internet technology, which no longer meets the new challenges of high-tech IT world.

Summarizing the spoken above, the following could be construed. Despite the obvious IP network concerns towards real-time traffic provision, the urgent obsolete IP by an alternative successor protocol does not seem a reasonable decision in the short term, since the vast majority of modern Internet applications are built on IP basis. At the same time, given fact solely means that IP remains the primarily underlying platform for application layer embedded in terminal network devices and supported by their operating systems. Eventually, there are no serious arguments to keep IP the only standardized interworking protocol as a unified packet delivery platform, [8], [9]. The next section outlines major principles of how to extend the autonomous system interoperability around ubiquitously deployed IP networks, intending the telephone network implantation into the body of next generation Internet.

Principles of IP network interoperability enhancement

The first principle of an enhanced IP network interoperability is reorganizing the AS-to-AS linkage by shifting it from IP layer to a sub-layer below, which conventionally designed for fast data switching in a local packet based network domain. This sub-layer is often referred to as L2.5 in terms of the OSI/ISO model, [2]. The L2.5 sub-layer functionally occupies an intermediate position between the data link layer (L2) and the network layer (L3), like known MPLS technology.

The second principle of an enhanced IP network interoperability is unification the two commonly counteracting mechanisms of data multiplexing under the joint protocol; these known mechanisms are circuit switching (CS) and packet switching (PS). According to the second principle, these two controversial types of data multiplexing will be embodied in a universal network device denoted as "Integrated Routing Switch" (IRS). On the first glance, the IRS entity looks very similar to known "Soft Switch"; though, IRS differs considerably in real-time data handling. The IRS claims an accurate TDM linkage provision, along with conventional IP-packet routing.

The third principle of an enhanced IP network interoperability is adaptation the layer L2 Ethernet technology to provide unified TDM/STM data link trunk on the edge of autonomous system, denote as "Unified Multiplexing Trunk" (UMT). The UMT will operate with the so called "Conveyor Transporting Modules" (CTM), originated in [10]. The CTM and UMT also look identical to the known synchronous transporting modules STM in SDH technology, but extend the statistical multiplexing functionality.

Conveyor transporting module technique for extended IP network interoperability

According the formulated above principles of an enhanced IP network interoperability, the new protocol data units (PDU) but not IP packets will circulate between any two adjacent autonomous systems (AS) in a converged NGN network. These PDUs look like modified Ethernet frames regular circulating in both directions in AS-to-AS linkage. Each one of those circulating frames bears conveyor-transporting module (CTM). This CTM contains two types of multiplexed data: the set of real time segments (i.e. encrypted voice data segments VDS) and the set of packet data (i.e. fragments of IP packets). The CTM modules, being encapsulated in modified Ethernet frames, circulate in synchronous mode at constant frequency determined by the physical link bandwidth along with adopted maximum Ethernet frame size, Fig.1.

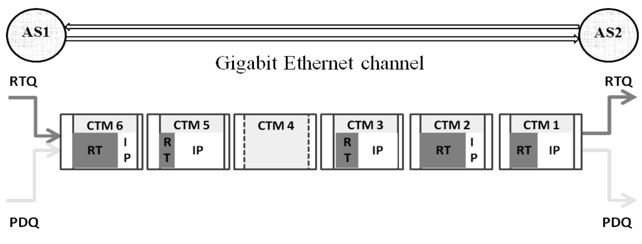

Fig.1 The modified Ethernet frame circulation

Unlike the known STM synchronous modules of SDH, the CTM conveyor modules have gradually variable capacity, depending on the intensity of incoming data stream. If there is no incoming data, the CTM module drops out in one or more circulation periods in order to save energy costs. The CTM more differs from

STM/SDH is that the CTM internal structure is dynamically changeable from any module to the next one. The real time segments may occupy any part of CTM (from 0% to 100%) whereas packet data take the rest CTM slot.

The algorithm of dynamic packet data multiplexing

The real time segments occupy CTMs upon TDM mechanism based on pre-established virtual connections with channel resource reservation; therefore, no real time data congestions occur. Instead, packet data scheduled for CTMs in best-effort mode to provide efficient throughput utilization. The packet data can be managed furthermore with generic integrated/differentiated service models (IntServ/DiffServ), [10].

The algorithm of CTM data multiplexing proceeds from the fact that, packet data slot of CTM has a variable size, changing from zero to maximum CTM capacity any time the current CTM module is scheduled. This fact makes it impossible regular allocation of diverse packets with random sizes, when desired tightly filling the momentary available CTM slot.

To overcome this issue, special method of packet data scheduling proposed in [10]. By this method, all IP packets, which had been addressed to the given output port of an Integrated Routing Switch (IRS), are arranged in a common packet data queue (PDQ) due to the assigned service policy and packet precedence, Fig.2. This PDQ is mapped onto concatenated byte sequence looking like formal grammar text, wherein distinct packets delimited by special commands of this grammar, like opening and closing brackets, Fig.2.

Imagine that current CTM module indicates L byte slot ready for packet queue scheduling. Now, the first L bytes be cut off the packet data queue (PDQ) and inserted in CTM. This cut fragment of PDQ is denoted as “Packet Data Block” (PDB). Each PDB may include either consistent IP packets or their fragments labeled by special commands of formal grammar, formed by combinations of a reserved byte (e.g. FF) and start/stop identifiers (e.g. FF01/FF02 for RTQ fragments, FF03/FF04 for PDQ ones). Reserved byte FF if occurred in fragment body is replaced by a combination of two bytes (e.g. FFFE). The size of resulting blocks is equal to the size of payload field of data link layer technology (Ethernet).

The receiving party of AS-to-AS linkage will concatenate consequently obtained CTM payload units to reconstruct the two separate digital flows, e.g. real-time segment queue (RTQ) and packet data multiplexed queue (PDQ), Fig.1. Next, the RTQ dropped into distinct real-time segments (RTS) sequence; these RTS are switched in TDM mode according the virtual circuit table (VCT) controlled by IRS. The PDQ is consequently dropped into distinct IP packets which handled by regular routing procedure (i.e. BGP4). Following aforesaid algorithm, the telephony traffic operates in TDM mode with accurate time delay control, whereas IP based Internet services function conventionally.

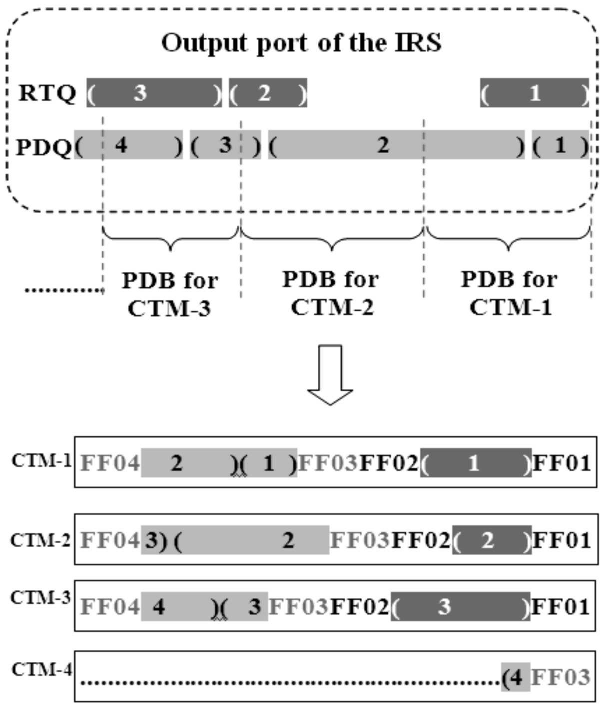

Fig.2 Real time and packet data scheduling in CTM

The Gigabit Ethernet internetworking linkage

Let's consider Gigabit Ethernet on the single-mode optical cable (IEEE 802.3z standard, physical layer specification 1000Base-LX, 10 microns fiber) to design an enhanced AS-to-AS linkage in aforesaid vision. The maximum length of such optical cable segment is 5000 m, [11]. The IEEE 802.3 frame comprises preamble (7 + 1 = 8 bytes), two MAC addresses (2 × 6 = 12 bytes), Type/Size field (2 bytes), payload of 46 Up to 1500 bytes, FCS checksum field (4 bytes); the inter frame gap (IFG) of minimum 12 bytes needed as well.

The minimum frame size is $8 + 12 + 2 + 46 + 4 + 12 = 84$ bytes; the maximum size (without IFG) is $8 + 12 + 2 + 1500 + 4 = 1526$ bytes, and the maximum frame cycle determined by $1526 + 12 = 1538$ bytes. Particular frame cycle and size depend on channel transmission rate.

The IEEE 802.3z Gigabit Ethernet differs from IEEE 802 in minimum frame option, which must be at least 512 bytes (no preamble and no IFG). The minimmum IEEE 802.3z frame is 512+8+12=532 bytes; the maximum length remains unchanged (1538 bytes). Calculate the min/max frequency of CTM cycling, which contained in IEEE 802.3z frames one per one. The Gigabit channel throughput calculated in byte per second is $10^9/8$. So, the min/max CTM transmission frequency is:

$$(f_{CTM})_{\min} = \frac{10^9\, byte / 8s}{1538 byte} \approx 81.3 KHz \quad ; \qquad (1)$$

$$(f_{CTM})_{\max} = \frac{10^9\, byte / 8s}{532 byte} \approx 235 KHz \quad . \qquad (2)$$

The CTM information capacity considers particular AS-to-AS Ethernet linkage with solely unique duo of negotiating parties (i.e. two adjacent border gate switching routers IRS). Therefore, no MAC addresses needed in such Ethernet linkage; so, two conventional MAC address fields of 12 bytes can be added to the payload field.

Now, the max size CTMs are transmitted at min frequency $(f_{CTM})_{min}$=81.3KHz (1); each CTM has 1500+12=1512 bytes. Again, the min size CTMs transmitted at max frequency $(f_{CTM})_{max}$=235KHz (2); each CTM has 512bytes–4bytes (FCS)–2 bytes (Type/Size field) = 506 bytes; this includes 12 bytes of MAC addresses and 494 bytes of the payload field. The aforesaid minimum CTM circulation frequency of 81.3 KHz in (1) far exceeds the commonly experienced voice sampling frequency, resulted in least voice latency of $(81.3\text{KHz})^{-1} \approx 0.01$ms; this merely affects the overall conversation quality over a packet based virtual circuit.

The max digital stream carried by CTMs at min circulation frequency and max Ethernet frame is

$$R_{\max} = 81.3 KHz \times (1512 \times 8) bit \approx 0.983 Gbit / s. \quad (3)$$

Suppose the max possible throughput of 0.800 GBit/s is fully dedicated for digital telephony transmission along with G729 voice codec of 8-times compression.

Take the standard E0 telephone channel of 64/8 = 8Kbits/s data streaming rate. These will result in max number of multiplexed voice connections carried by IEEE 802.3z Gigabit Ethernet link: $(0.8 \bullet 10^9 \times \text{bit/s}) / (8 \bullet 10^3 \times \text{bit/s}) = 10^6$. So, about 100’000 QoS based telephone connections can be supported at the peak load on the IEEE 802.3z; herewith, max latency takes 50 ms plus physical signal propagation time.

In contrast to conventional TDM circuits, the enhanced method of AS-to-AS Ethernet linkage presented above, benefits the more accurate signaling procedure with comprehensive resource reservation, e.g. the TDM dedicated channel capacity of an AS-to-AS linkage does not actually occupied until the signaling process finishes and real time flow starts coming to switching device (i.e. IRS) of a converged packet based network. Because of that, any temporally unused resources are dynamically scheduled for packet traffic delivery.

Conclusion

The issues of telephony to Internet convergence considered. Comparative analysis given on alternative trends in telecoms related to circuit and packet switching. Noted, that constrains of TCP/IP hinders NGN confluence, so new researches needed on network system architecture.

Traditional telecoms and Internet service providers permanently compete for IT market niches having their pros and cons. The circuit switched networks provide time division multiplexing to benefit constant bit rate, still suffering monopole channel occupation and superfluous billing. In turn, packet switching rides statistical multiplexing to low cost data exchange with no much care about momentary throughput faults.

As vast IP-applications flood the market, no urgent successor-protocol reasonable in short term, despite IP concerns when real-time traffic. Yet, it solely remains IP primary underlay for customer application. Thus, not much pro-IP solo claims for interworking performance.

Basic principles introduced to enhance autonomous network interoperability. A novel mechanism of conveyor transporting modules originated for adjacent autonomous systems linkage to benefit packet and circuit switching unification. An alternative algorithm proposed for dynamic packet data allocation in the unified network-to-network link. The Gigabit Ethernet linkage outlined for diverse network convergence based on extended interoperability solution.

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Тихонова О.В.

Метод розширення можливостей взаємодії елементів конвергентної мережі, що базується на технології Ethernet

Проблематика. Уніфікація комунікаційних послуг на базі протоколу IP часто стикається з проблемою забезпечення гарантій QoS при передачі трафіку в режимі реального часу. Втрати пакетів та випадкові затримки в статистично ущільненому потоці даних, очевидно, мають негативний вплив при дистанційному аудіо/відео-спілкуванні або у разі міжмашинної взаємодії в режимі реального часу. Відомі підходи до вирішення цього питання, що базуються на протоколі IP, не можуть задовольнити високі вимоги операторів та споживачів до якості послуг; це стримує конвергенцію мережі Інтернет та телефонних мереж з точки зору якості телекомунікаційних послуг. Тому для ефективного вирішення цих проблем потрібні додаткові дослідження.

Мета досліджень. Дана стаття спрямована на розширення можливостей взаємодії елементів конвергентної телекомунікаційної мережі для забезпечення якості послуг при передачі даних у режимі реального часу.

Методика реалізації. Аналіз поточного стану створення конвергентних мереж типу "Інтернет + телефонна мережа", а також синтез вдосконаленого інтерфейсу між-мережевої взаємодії для гетерогенної мережі.

Результати досліджень. Сформульовані основні принципи для розширення сумісності автономних мережевих систем. Представлено механізм конвеєрних транспортних модулів для конвергентної мережі з метою поєднання достоїнств, притаманних технологіям комутації пакетів та комутації каналів.

Висновки. Традиційні постачальники телекомунікаційних та Інтернет-послуг постійно конкурують за ніші IT-ринку, маючи при цьому свої плюси і мінуси; при цьому на ринку існує велика кількість рішень заснованих на протоколі IP. Тому заміна цього протоколу іншим не є виправданою у короткостроковій перспективі. З огляду на це, в роботі запропоновано альтернативний алгоритм для динамічного розподілу пакетних даних в уніфікованому між-мережевому каналі зв'язку, який розширює можливості між-мережевої взаємодії у конвергентній мережі. Базуючись на запропонованому алгоритмі, розроблено між-мережевий стик для каналу Gigabit Ethernet.

Ключові слова: між-мережева взаємодія; QoS; додатки реального часу.

Тихонова Е.В.

Метод расширения возможностей взаимодействия элементов конвергентной сети, основанной на технологии Ethernet

Проблематика. Унификация коммуникационных услуг на базе протокола IP часто сталкивается с проблемой обеспечения QoS при передаче трафика в режиме реального времени. Потери пакетов и случайные задержки в статистически мультиплексированном потоке данных, очевидно, имеют негативное влияние при дистанционном аудио/видео-общении или в случае межмашинного взаимодействия в режиме реального времени. Известные подходы к решению этого вопроса, основанные на протоколе IP, не могут удовлетворить высокие требования операторов и потребителей к качеству услуг; это сдерживает конвергенцию сети Интернет и телефонных сетей с точки зрения качества телекоммуникационных услуг. Поэтому для эффективного решения этих проблем нужны дополнительные исследования.

Цель исследований. Данная статья направлена на расширение возможностей взаимодействия элементов конвергентной телекоммуникационной сети для обеспечения качества услуг при передаче данных в режиме реального времени.

Методика реализации. Анализ текущего состояния создания конвергентных сетей типа "Интернет + телефонная сеть", а также синтез усовершенствованного интерфейса межсетевого взаимодействия для гетерогенной сети.

Результаты исследований. Сформулированы основные принципы для расширения совместимости автономных сетевых систем. Представлен механизм конвейерных транспортных модулей для конвергентной сети с целью объединения достоинств, присущих технологиям коммутации пакетов и коммутации каналов.

Выводы. Традиционные поставщики телекоммуникационных и Интернет-услуг постоянно конкурируют за ниши IT-рынка, имея при этом свои плюсы и минусы; при этом на рынке существует большое количество решений, основанных на протоколе IP. Поэтому замена данного протокола не оправдана в краткосрочной перспективе. Учитывая это, в работе предложен альтернативный алгоритм для динамического распределения пакетных данных в унифицированном межсетевом канале связи, который расширяет возможности межсетевого взаимодействия в конвергентной сети. На базе предложенного алгоритма разработан межсетевой стык для канала Gigabit Ethernet.

Ключевые слова: межсетевое взаимодействие; QoS; приложения реального времени.