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VoIP Packet Delay Techniques : A Survey

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Abstract- The continuous development in the field of communication have paved the way for Voice over Internet Protocol (VoIP). VoIP is a group of hardware and software that facilitates people to utilize the Internet as the transmission medium for telephone calls by transmitting voice data in packets using IP instead of using conventional circuit transmissions of the Public Switched Telephone Network (PSTN). At present, VoIP is becoming an important tool for quick communication across the world. There are several Internet telephony applications existing at present. The major disadvantage in VoIP is that the packet delay. In VoIP, the terminology jitter is used to refer the type of packet delay where the delay has a huge setback in the quality of the voice conversation. Several packet delay techniques were proposed in recent years. Some of the important packet delay techniques are discussed in the literature. This survey would definitely help the researchers to carry out their research for providing better communication in VoIP without any delay.

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VoIP Packet Delay Techniques : A Survey

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I. Introduction

There are numerous attractive alternatives both to conventional public telephony and to charter lines for networking in these days. Among the most remarkable are networking technologies based on various kinds of voice transmission called Voice over IP (VoIP). Voice over IP (VoIP) refers to real-time delivery of packet voice across networks utilizing the Internet protocol. VoIP's request is based on its capability to make possible for voice and data convergence at an application layer [1, 2].

One of the classic problems with the accomplishment of packet voice over IP is the complexity of QoS guarantee. Voice quality is affected by delay, delay jitter and unreliable packet delivery all of which are usual characteristics of the essential IP-network service [3]. IP traffic is obviously treated as "best effort" and transmits on a first-come, first-served basis.These characteristics have supplied to large delays and large delay variations in packet delivery, which are the most significant concerns of packeted voice QoS requirement.

This technology provides new opportunities for the growth of new applications and educational services, chiefly through the potential for converging Voice with supplementary media and data. In the long-term, VoIP is expected to impact on some of the better developments in higher education, for instance, virtual universities, personality management and incorporation with enterprise-level services and applications [4].

The fundamental process carried out in a VoIP call is as given below [5]:

1. Transformation of the caller's analog voice signal into a digital format.
2. Compression and transformation of the digital signal into discrete Internet Protocol packets
3. Transmission of the packets through the Internet or other IP-based network
4. Reverse transformation of packets into an analog voice signal for the call recipient.

Recently, huge numbers of organizations, public and private, have begun evaluating IP technologies as they believe that IP-based systems present increased reliability and fault-tolerance. This is the primary stage of the creation of the 'converged network' in which a single network replaces the current set-up of twin, separate networks of voice (PBX) and data (LAN) [6]. This network possibly will be an organization's internal LAN, a leased network, the PSTN or the open Internet [7]. The compression process is carried out by a codec, a voice-encoding algorithm, which permits the call to be sent over the IP network inside the network's accessible bandwidth.

Packet delay is perhaps the most difficult constituent of network behavior to examine—with loss, for instance, the packet either shows up at the receiver or it does not, while with delay there are several shades of possibility and meaning in the time necessary for a packet to arrive. Similarly, delay variation is potentially the better source of information regarding the VoIP network, as one of the principle elements contributing to delay is queuing within the network, which is of vital importance in understanding how network capacities evolve over time [8].

Several packet delay techniques were proposed in recent years. Some of the important techniques are discussed in the following section.

II. Literature Survey

The difficulty of multiple-packet bundling to enhance spectral effectiveness in cellular networks is examined. The packet size of real-time data, like Voice over Internet Protocol (VoIP), is frequently very small. On the other hand, the general use of Time-Division Multiplexing (TDM) limits the number of VoIP users

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supported as a packet has to remain until it gets a time slot, and if only one small VoIP packet is placed in a time slot, capacity is wasted. Packet bundling can ease that problem by sharing a time slot between multiple users. A current revision of cdma2000 1xEV-DO has established the concept of the Multiuser Packet (MUP) in the downlink to conquer restrictions on the number of time slots. But, the effectiveness of packet bundling is not well understood, mainly in the occurrence of time-varying channels. Baek-Young Choi et al., [9] proposed a novel quality-of-service (QoS) and channel-aware packet bundling algorithm that takes advantage of adaptive modulation and coding. It reveals that optimal algorithms are Nondeterministic Polynomial time (NP)-complete, suggest heuristic approaches, and utilize analytical performance modeling to demonstrate the gains in capacity that can be attained from the packet bundling algorithms. It also reveals that channel utilization can be considerably increased by slightly delaying some real-time packets within their QoS requirements while bundling those packets with like channel conditions. It is authenticated through extensive OPNET simulations with an entire evolution-data optimized (EV-DO) implementation.

Yensen et al., [10] proposed a new algorithm for predicting audio packet play out delay for voice conferencing applications that use silence suppression. The proposed algorithm utilizes a Hidden Markov Model (HMM) to envisage the play out delay. Numerous existing algorithms are reviewed to show that the HMM technique is based on a combination of different desirable features of other algorithms. Voice over Internet protocol (VoIP) applications generates packets at a deterministic rate but different queuing delays are added to the packets by the network causing packet interarrival jitter. Playout delay prediction techniques lists audio packets for playout and challenge to make a reasonable compromise between the number of lost packets, the one-way delay and the delay variation as these criteria cannot be optimized concurrently. Particularly, the proposed HMM technique is revealed to make a good compromise among the mean end-to-end delay, end-to-end delay standard deviation and average packet loss rate.

James et al., [11] introduced voice over IP networks and services in ways that satisfy the voice quality expectations of our customers, we have been conducting laboratory studies of how VoIP transmission affects voice quality while also carefully monitoring and managing several field implementations of VoIP. This article reviews and gives a helpful progress report on the industry's development to VoIP. It also review the data on the voice quality consequences of packet loss, delay, speech coders, packet loss concealment algorithms, and the compression choice of suppressing transmission through silence. Since the common problem of echo has emerged frequently in the VoIP environment, it reviews this issue in a few details. Packet loss and delay variation sizes prepared on private VoIP networks are evaluated, and the data here are persuaded. It completes this case by making that the network planning tool recognized as the E-model is presently an inexact predictor of VoIP network presentation.

The widespread usage of mobile devices in the IP network has guide to a new attempt to pertain a Power-Saving Mode (PSM) to real-time traffic like Voice over IP (VoIP). Hyun-Ho Choi et al., [12] evaluated the performance of the PSM when the PSM is used for VoIP services of mobile devices. Obtaining the performance of each conversational party into explanation, it takes two different kinds of PSMs: one is employed through the talk-spurt periods and the other is employed through the mutual silence periods of two conversational parties. The presentation of each PSM is examined with respect to buffering delay, the possibility of packet drop, and power consumption of a mobile VoIP device. After that, the maximum bound of sleep interval in every period is gained, which reduces the power consumption of the mobile device avoiding violations of the Quality-of-Service (QoS) of VoIP. In the various network environments, the proposed PSM for VoIP considerably reduces the power consumption while fulfilling the end-to-end delay and packet drop probability restrictions of a VoIP connection.

Recently, there has been a remarkable increase in the popularity of VoIP services as the outcome of large growth in broadband access. The same Voice-over-Internet protocol (VoIP) service creates new challenges when arranged over a wireless mesh network, at the time of enabling users to create voice calls using WiFi phones. Owing to interference Packet losses and delay in a multiple-hop mesh network with limited capacity, it can considerably humiliate the end-to-end VoIP call quality. It converse about the fundamental requirements for efficient deployment of VoIP services for the mesh network. Ganguly et al., [13] presented and evaluated practical optimizing techniques that can enhance the network capacity, maintain the VoIP quality and handle user mobility efficiently. A real testbed and ns-2 provides insights into the presentation issues and reveals the level of enhancement that can be achieved by the proposed techniques. Particularly, the packet aggregation is discovered with the help of header compression that can raise the number of supported VoIP calls in a multihop network by 2-3 times. The proposed fast path switching is extremely efficient in preserving the VoIP quality. This fast handoff scheme attains almost unimportant disruption during calls to roaming clients.

VoIP (Voice over Internet protocol) technology has rapidly been increasing recently, which transmits voice packets by utilizing the User Datagram Protocol (UDP). VoIP quality is very hard to expect because it is

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difficult to envisage the influence of packet delay, packet lose, packet error, etc. Bih-Hwang Lee et al., [14] proposed an embedded call admission control (CAC) mechanism by applying real-time transfer protocol (RTP) and the real-time control protocol (RTCP) for VoIP services over hybrid fiber/coaxial (HFC) networks. The proposed CAC mechanism is estimated by the impact of the different traffic loads in cable modem termination system (CMTS), which calculates how VoIP quality satisfies the user's requirements under different restrictions on cable networks. It confers about VoIP CAC mechanism for the upstream channel according to the Data over Cable Service Interface Specifications (DOCSIS) version 1.1 and chiefly considers G.723.1 voice packets at the transmission rate of 6.3 kbps. The performance size of the proposed embedded CAC mechanism is achieved under different network constrains, which consists of throughput, packet dropping ratio and call blocking ratio. It clearly gives effectiveness and fast method for CMTS to decide how many calls can be allowed.

It is recognized that measuring the one-way delay of Voice-over-IP (VoIP) packets is an intimidating task. The confront lies in the fact that the Internet implicitly implements the end-to-end principle. This means that the endpoints are anticipated to operate separately. Exclusive of a synchronized timing, it is hard for an endpoint to gauge the one-way delay. Ngamwongwattana et al., [15] presented the refined version of the novel VoIP measurement methodology called Sync & Sense of Periodic Stream that can overcome such a challenge. Sync & Sense is single in that it can virtually coordinate the transmission and reception timing of the VoIP session without the need of a synchronized clock. Therefore it reveals that Sync & Sense can accurately gauge the one-way network delay of the VoIP packets (without propagation delay). While time skew is very general in any system involving a clock, we the question arises on how Sync & Sense can deal with the time skew without the need of a synchronized clock.

In the context of the IEEE 802.16e standard, a Dual Power-Saving Mode (DPSM) algorithm for Voice over IP (VoIP) traffic whose voice codec supports voice activity detection is proposed by Lee et al., [16]. The proposed algorithm uses the indolence of the voice codec of every conversing party through mutual silence periods. Utilizing the suggested method, the length of the sleep intervals differs during mutual silence periods, while during talk-spurt periods it is permanent according to the VoIP packet generation ratio. The presentation of the DPSM algorithm for the average packet-buffering delay in the Base Station (BS) and the energy consumption of a Mobile Station (MS) is estimated numerically and authorized with the help of computer simulation. Therefore it shows that when the proposed combined method is used, the energy consumption of an MS is significantly less when a PSM that only uses sleep intervals of a permanent length is operating. This development in performance comes at the cost of better packet-buffering delay in the BS.

Capacity has been a significant matter for several wireless backhaul networks. The multihop nature and the huge per packet channel together access overhead which can guide to its low channel effectiveness. The problem may receive even badly when there are numerous applications transmitting packets with small data payloads, for example, Voice over Internet protocol (VoIP). Earlier, the utilization of several parallel channels and employing packet concatenation were taken care as independent solutions to these problems. On the other hand, there are unavailable work on the integrated design and performance study of a complete scheduler architecture joining these two schemes. Wei-chih Hong et al., [17] proposed a scheduler that concatenates small packets into large frames and sends them through multiple parallel channels with an intelligent channel selection algorithm between neighboring nodes. In addition the predictable capacity improvements, also obtain delay bounds for this scheduler. Depending upon the delay bound formula; Call Admission Control (CAC) of a wide variety of prepared algorithms can be achieved. It reveal the important capacity and resequencing delay improvements of this novel design with a voice-data traffic mixing instances, by means of both numerical and simulation results. It is revealed that the proposed packet concatenation and channel selection algorithms largely perform well in the round-robin scheduler in a multihop scenario.

TCP has usually been considered unsuitable for real-time applications. Nevertheless, popular applications like Skype use TCP since UDP packets cannot exceed through restrictive Network Address Translators (NATs) and firewalls. Encouraged by this observation, it examines the delay performance of TCP for real-time media flows. Brosh et al., [18] developed an analytical performance model for the delay of TCP. Some of the experiments are conducted to authenticate the model and to calculate the impact of various TCP mechanisms on its delay performance. Therefore it specifies that simple application-level schemes, like packet splitting and parallel connections, can decrease the delay of real-time TCP flows by as much as 30% and 90%, correspondingly.

Voice communications e.g., telephony are delayed responsive. Existing Voice-over-IP (VoIP) applications convey voice data in packets of very small size to reduce packet delay, causing very incompetent use of network bandwidth. Sze et al., [19] proposed a multiplexing scheme for improving the bandwidth efficiency of existing VoIP applications. By fixing a multiplexer in an H.323 proxy, voice packets from many sources are joined into single IP packet for transmission.

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The receiver-end proxy returns the innovative voice packets of the de multiplexer before sending them to the end-user applications. The multiplexing scheme is completely well-matched with existing H.323-compliant VoIP applications and can be voluntarily deployed.

Speech communication using the Voice over Internet Protocol (VoIP) is very frequent today. The fundamental network channel may be the Public Switched Telephone Network (PSTN channel), satellite channels or cellular wireless channels to mention a few. The packetization of speech and its transmission through packet switched networks, however, initiate numerous impairments for example delay, jitter, packet loss and decoder clock offset, which disgrace the quality of the speech. Ogunfunmi et al., [20] presented an overview of the challenges and a description of the advanced signal processing algorithms used to combat these impairments and render the perceived quality of a VoIP conversation to be as good as that of the existing telephone system. An instance of a speech coder is also designed for packet-switched networks and converses the possibilities for hardware implementations.

Satellites are predicted to be balancing to prospect terrestrial networks in deploying multimedia communication systems. The use of geostationary multibeams and on-board processing gives a great chance for the immediate deployment of real time services such as IP telephony services over satellites. The effects of IP telephony move over satellite channels have not been examined in detail. The Cruickshank et al., [21] presented an overview of the VIP-TEN project architecture and the VoIP measurement campaign over the EuroSkyWay test-bed. The geostationary satellites can take VoIP traffic and present a good quality service in terms of packet loss and jitter, and average to poor quality in terms of packet delay. It also studies the delays in setting/joining audio conferences and multicast group organization over satellites.

In IEEE 802.11e-based WLANs, link adaptation mechanisms, which select the transmission rate of each node, aggravate unexpected and random variations on the efficient channel capacity. When these alterations are towards lower bit-rates, inelastic flows, like VoIP, it can undergo from sudden congestion, which outcomes higher packet delays and losses. A VoIP codec selection algorithm has been proposed by Sfairopoulou et al., [22] as a solution to this issue, which is triggered both by channel rate changes as well as in combination with a call admission control mechanism. An important enhancement in terms of hotspot capacity for VoIP calls can be attained by selecting the VoIP codec adaptively in a multi-rate scenario. By describing a new grade of service-related parameter, Qmacr, which captures the tradeoff among dropping and blocking probabilities and professed speech quality, the codec selection algorithm can be tuned to reach maximum capacity without strictly penalizing any of those variables, and therefore satisfying both technical and user quality requirements.

Considering voice as a leading telecommunication service, the performance of Voice over IP (VoIP) plays an important role in deployment of worldwide interoperability for microwave access (WiMAX) technology giving all-IP network services. Lin et al., [23] investigated the performance of a WiMAX-based VoIP established under the mobile Taiwan (M-Taiwan) field-trial funded program. To gain the objectives of the trial the measurement results are expressed in the form of Mean Opinion Score (MOS), packet loss, packet delay and jitters. For the worst-case scenario, the analyses were performed under a tough condition of both communicating devices, wirelessly associated to the same WiMAX base station under a heavy background traffic and interference. Ahead of the analysis, the field measurements verify an excellent performance when both communicating devices kept stationary and demonstrate an acceptable quality for the service when both communicating devices are on the move at a speed of 50 km/h.

The prologue of the IP multimedia subsystem on 3G cellular networks and the combination with other widely deployed wireless networks based on the IEEE 802.11 protocol family necessitate support for both mobility and quality of service. When mobile systems go across heterogeneous networks, ongoing real-time sessions are affected not only by handoff delay but also by various packet delay and bit rate. Bernaschi et al., [24] proposed a cross-layer mechanism that takes into account mobility at different layers of the network stack in order to yield better quality for VoIP, videoconferencing, and other real-time applications. Finally it expresses the cross-layer architecture, adaptation techniques and prototype implementation.

Table1 : An Overview of the Existing VoIP Packet Delay Techniques

Method	Technique Used
Baek-Young Choi et al., [9]	Novel Quality-of-Service (QoS) and channel-aware packet bundling algorithm
Yensen et al., [10]	New algorithm for predicting audio packet playout delay
Ganguly et al., [13]	Evaluated practical optimizing techniques
Bih-Hwang Lee et al., [14]	Embedded Call Admission Control (CAC) mechanism
Ngamwongwattana et al., [15]	Refined version of the novel VoIP measurement methodology
Lee et al., [16]	A Dual Power-Saving Mode (DPSM) algorithm
Brosh et al., [18]	Developed an analytical performance model

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Sze et al., [19]	Multiplexing scheme
Ogunfunmi et al., [20]	An overview of the challenges and a description of the advanced signal processing algorithms
Bernaschi et al., [24]	A cross-layer mechanism

III. Problems And Directions

Although the above discussed techniques have practically provided some method to reduce the packet delay, but there is still no effective technique to completely reduce the packet delay. The following points should be taken into account for better communication without any delay.

- The major reason for packet delay is congestion. During the process of communication, several packets are delayed due to congestion in the network. So, in future, researchers should consider congestion as a serious problem and propose some technique to handle this congestion.
- The VoIP is not only loss-adaptive but also delay-adaptive. Specifically, they can have either a constant time, or adapt to changes in packet delay and fix the time accordingly. Such adaptive technique can keep track of packet delays, and reflect any change in packet delays.

IV. Conclusion

VoIP is presented on many smartphones and Internet devices so that users of portable devices may use phones, may set calls or send SMS text messages over 3G or Wi-Fi. The delay is identified from the start of the packet from which it is transmitted at the source to the end of the packet and then received at the destination. A constituent of the delay which does not differ from packet to packet can be unnoticed; hence if the packet sizes are equal and packets always obtain the same time to be processed at the destination then the packet arrival time at the destination could be utilized instead of the time the end of the packet is received. For future work, the above techniques plans to investigate the presentation of the packet delay in the VoIP applications.

References Références Referencias

1. Jane Dudman and Gaynor Backhouse, "Voice over IP: what it is, why people want it, and where it is going", JISC Technology and Standards Watch, 2006.
2. Jonathan Lennox, Henning Schulzrinne, "Feature Interaction in Internet Telephony",Columbia University, 2000.
3. Federal Communications Commission, "Voice over Internet Protocol" <http://www.fcc.gov/voip/>.
4. Hazem M. El-Bakry and Nikos Mastorakis, "Fast Detection of Specific Information in Voice Signal over Internet Protocol", Proc. of the 7th WSEAS Int. Conf. on Computational Intelligence, Man-Machine Systems and Cybernetics, Pp. 125-136, 2006.
5. Greg S. Tucker, "Voice Over Internet Protocol (VoIP) and Security", SANS Institute Reading Room site, 2005.
6. Mohamed, A., 2006. Converged Networks: The VoIP revolution. Computer Weekly. 12 September 2006.
7. Gradwell, P. 2006. VoIP Discussion at the BCS Specialist Internet Group. British Computer Society: London. Available at: http://www.bcs.org/ server. php?show=ConWebDoc.3556.
8. S. Saravanan and E Karthikeyan, "A Protocol to Improve the Data Communication Over Wireless Network", International Journal of Wireless & Mobile Networks (IJWMN) Vol. 3, No. 5, Pp. 95-112, 2011.
9. Baek-Young Choi; Jung Hwan Kim; Beard, C., "QoS and Channel-Aware Packet Bundling for Capacity Improvement in Cellular Networks" , IEEE Transactions on Vehicular Technology, Vol.59, No.8, Pp. 3886-3901, 2010.
10. Yensen, T.; Lariviere, J.P.; Lambadaris, I.; Goubran, R.A., "HMM delay prediction technique for VoIP", IEEE Transactions on Multimedia, Vol.5, No.3, Pp. 444- 457, 2003.
11. James, J.H.; Bing Chen; Garrison, L., "Implementing VoIP: a voice transmission performance progress report", IEEE Communications Magazine, Vol.42, No.7, Pp. 36- 41, 2004.
12. Hyun-Ho Choi; Jung-Ryun Lee; Dong-Ho Cho, "On the use of a power-saving mode for mobile VoIP devices and its performance evaluation", IEEE Transactions on Consumer Electronics, Vol.55, No.3, Pp. 1537- 1545, 2009.
13. Ganguly, S.; Navda, V.; Kyungtae Kim; Kashyap, A.; Niculescu, D.; Izmailov, R.; Sangjin Hong; Das, S.R., "Performance Optimizations for Deploying VoIP Services in Mesh Networks", IEEE Journal on Selected Areas in Communications, Vol.24, No.11, Pp. 2147- 2158, 2006.
14. Bih-Hwang Lee; Jhih-Ming Chen, "VoIP services with embedded CAC over cable modem systems", IEEE Transactions on Consumer Electronics, Vol.51, No.3, Pp. 815- 823, 2005.
15. Ngamwongwattana, B.; Thompson, R., "Sync & Sense: VoIP Measurement Methodology for Assessing One-Way Delay Without Clock Synchronization", IEEE Transactionson Instrumentation and Measurement, Vol.59, No.5, Pp. 1318-1326, 2010.
16. Lee, J.-R.; Cho, D.-H., "Dual power-saving modes for voice over IP traffic supporting voice activity detection", IET Communications, Vol.3, No.7, Pp. 1239- 1249, 2009.

17. Wei-chih Hong; Zsehong Tsai, "A Multichannel Scheduler for High-Speed Wireless Backhaul Links with Packet Concatenation", IEEE Transactions on Mobile Computing, Vol.9, No.2, Pp. 201- 214, 2010.
18. Brosh, E.; Baset, S.A.; Misra, V.; Rubenstein, D.; Schulzrinne, H., "The Delay-Friendliness of TCP for Real-Time Traffic", IEEE/ACM Transactions on Networking, Vol.18, No.5, Pp. 1478- 1491, 2010.
19. Sze, H.P.; Liew, S.C.; Lee, J.Y.B.; Yip, D.C.S., "A multiplexing scheme for H.323 voice-over-IP applications", IEEE Journal on Selected Areas in Communications, Vol.20, No.7, Pp. 1360- 1368, 2002.
20. Ogunfunmi, T.; Narasimha, M.J., "Speech over VoIP Networks: Advanced Signal Processing and System Implementation", IEEE Circuits and Systems Magazine, Vol.12, No.2, Pp. 35- 55, 2012.
21. Cruickshank, H.; Sun, Z.; Carducci, F.; Sanchez, A., "Analysis of IP voice conferencing over EuroSkyWay satellite system", IEE Proceedings-Communications, Vol.148, No.4, Pp. 202- 206, 2001.
22. Sfairopoulou, A.; Bellalta, B.; Macian, C., "How to tune VoIP codec selection in WLANs?" IEEE Communications Letters, Vol.12, No.8, Pp. 551-553, 2008.
23. Lin, Y.-B.; Sung, Y.-C.; Rashvand, H.F.; Liu, C.-L.; Liao, Y.-J., "Mobile-Taiwan experience in voice over IP-worldwide interoperability for microwave access trial", IET Communications, Vol.4, No.9, Pp. 1130-1141, 2010.
24. Bernaschi, M.; Cacace, F.; Iannello, G.; Vellucci, M., "Mobility Management for VoIP on Heterogeneous Networks: Evaluation of Adaptive Schemes", IEEE Transactions on Mobile Computing, Vol.6, No.9, Pp. 1035- 1047, 2007.