Asian Journal of Information Technology 6 (5): 589-594, 2007

TCP Friendly Equation Rate Control Based Cryptographic Active Voice Transcoding (CAVT-ER) Congestion Control Mechanism for Voice Over IP (VOIP) Transmission-A Case Study

[1]P. Bhuvaneswari and [2]A.M. Natarajan
[1]Department of Information Technology, Institute of Road and Transport Technology, Erode-638 316, Tamilnadu, India
[2]Kongu Engineering College, Perundurai-638 052, Tamilnadu, India

Abstract: The major performance limitation for voice communication over an IP network is the inefficient use of network resources such as CPU and Bandwidth. The increase in the growth of an Internet increases the voice transmission that degrades the quality of real time voice communication due to packet overhead, network congestion and improper queuing in an IP network. These overheads affect the overall network performance and Qos parameters of voice communication such as packet loss, packet delay and packet latency or jitter. In our study, we have discussed how the active networking services are suited to reduce the impact of congestion in an IP network. An analysis of how the TCP Friendly Equation Rate Control based Cryptographic Active Voice Transcoding (CAVT-ER) Congestion Control mechanism should adapts the throughput generated by the voice packets in response to congestion while maintaining a steady voice data throughput in order to achieve effective QoS of VoIP applications where a smooth sending rate is of importance. We discuss the analytical results of the CAVT-ER performance. As a case study we show that our proposed algorithm would directly control congestion in the network and indirectly lead to fast bandwidth adaptation and minimize packet loss, packet delay and jitter.

Key words: Voice over IP (VoIP), quality of service, packet loss, packet delay, jitter, transcoding, cryptography, congestion, security, authentication

INTRODUCTION

The convergence of data communications (packet switched) and voice-based communications (traditionally circuit switched) into single IP-based core architecture offers an opportunity for large savings in communication cost. The audio codecs are used for VoIP flow transmission, which are operated based on advanced voice-compression techniques, can be used to generate low bit data rates (less than 10Kbps) and bandwidth is consumed only when voice packets are delivered. Delay or latency sensitive traffic such as voice and audio has unacceptable performance if long delays are incurred. A bounded delay on the voice delivery can be achieved either by restricting the offered load or by adding QOS mechanisms to the Internet. The major performance limitations for voice communication over an IP network is the inefficient use of network resources such as buffers and bandwidth due to packet overhead, network congestion, improper queuing and hence proposes improvements. We propose a new TFRC based congestion control scheme for voice traffic called CAVT-ER. In this scheme, the network participates in controlling its traffic rate and bandwidth utilization dynamically in order to ensure acceptable performance for real time voice communication. i.e., CAVT-ER is applied at an active node or at active routers. It also adjusts the bandwidth consumption of voice packets transmitted between 2 end systems to reduce an impact of congestion on the network. And also we present an architecture that is working based on the active networking services. Then the analytical results from the case study shows that our CAVT-ER scheme is different from already existing adaptive rate control mechanisms and acts as an effective congestion control mechanism.

MATERIALS AND METHODS

This study describes active networking concepts and its related issues, impact on VoIP transmission, Qos parameters for VoIP.

Active networking concepts and its related issues: The active networking services is an emerging trend, which

Corresponding Author: P. Bhuvaneswari, Department of Information Technology, Institute of Road and Transport Technology, Erode-638 316, Tamilnadu, India

reduces the burden on the end systems and its direction is the movement of end or edge based QOS control schemes to inside the network to solve the problems encountered in end-based implementations.

Active Networking Services (ANS): Defense Advanced Research Projects (DARPA) research community, as a future direction of networking system (Tennenhouse *et al.*, 1997) introduced the concept of active networks in 1994. In addition to packet forwarding mechanism in traditional IP network, active networks allow the network nodes to perform user or application specific computations on user data passing through them. In active networks, normal packets are replaced with active packets containing small programs and possibly data. Active network technology can be used as a useful tool for providing dynamic or on demand QoS where active nodes can adjust their QoS configuration as instructed by the programs in active packets.

Integrated Service (IntServ) model described in RFC 1633 (Braden *et al.*, 1994; Detti *et al.*, 1999; Bernet *et al.*, 2000) provides resource sharing mechanism which could be requested dynamically per-flow. The model deploys resource reservation mechanism in which network resources are reserved along each communication path. However, this model suffers from its scalability problem due to the overhead on maintaining the path state information at each node or router. Rather than providing per flow QoS, DiffServ provides coarse grained QoS for traffic aggregates or classes and applies different per-hop services to different classes. These works share the idea on deploying IntServ at the edge and Diff- Serv at the core networks. They introduce the use of a signaling mechanism in IntServ to communicate the QoS requirements to DiffServ. Hence, they cannot really provide QoS on demand, which could be requested directly by authorized users.

The Active Networking Services (ANS) can provide Qos on demand qualities and functionalities by using the store-compute-and-forward concept. Therefore, in addition to the usual payload to provide computation ability that packet can carry small chunks of code that can be executed in the active nodes for extra processing functionality. The ANS is a promising service and performance booster for future networks. The ANS should also emphasize its advantages while tackling the issues of complexity and security. Its direction is the movement of an end or edge-based Qos Control schemes to inside the network to solve the problems encountered in end-based implementations (Yan and Mabo, 2004). This approach can outperform end-to-end solutions in non-active networks for the following 3 reasons:

- First, active network employs a packet centric approach in which the packets can contain custom code and in which the code is executed at the intermediate nodes as the packets travel through the network.
- Second, an active network reduces the network traffic. In non-active environment, algorithms for traffic control must be applied uniformly to a packet flow from an entry point in a network, toward the edge of the network where the users are, while in an active network, traffic control is only performed through selected active routers.
- Third, an active network speeds up the deployment of new services.

Tansupasiri and Kanchanasut (2003) proposed another concept of using active networks to provide dynamic QoS. In (Tansupasiri and Kanchanasut, 2003), D-QoS system automatically adjusts its QoS settings according to user requests. In this study, CAVT-ER scheme is to be applied at any intermediate node or at an active node which automatically adjusts the congestion level in the network without getting the user requests, reduces the network traffic, improves the QOS parameters of voice communication and leads to fast bandwidth adaptation.

ANS based loss control techniques: The commonly employed techniques are network layer error control and reencoding, traffic filtering, traffic dispersal and retransmission. The traffic filtering technique can be further divided into two types: Intelligent packet discarding and transcoding (Yan and Mabo, 2004).

Transcoding: The transcoding can be used to transform the user data within the network to conform to network conditions either by changing its voice coding format, altering coding parameters or transforming prioritized data streams to non-prioritized streams (Yan and Mabo, 2004).

Impact on VOIP transmission

Packet header overhead: The potentially negative effects of protocol header overhead, which is twice the voice payload generated by the high compression audio codecs. It increases the network traffic.

Network congestion: Congestion is the condition reached when the demand for network resources exceeds the available resources for an extended interval of time. Congestion causes excessive delay, loss or both.

Improper queuing: Queuing is a service discipline or packet scheduling scheme used to manage the voice packets flow in an IP network. The primary function of the queuing methods is to allow for the reservation of network bandwidth, so that each individual flow can be ensured a specified share of bandwidth at each traversed node along the path. The uses of unsuitable queuing mechanisms make the voice traffic flow over an IP network more complex.

QOS parameters for VOIP:

Packet loss: It refers to packets that do not arrive from the sender to the receiver. Packet loss can significantly degrade the quality of voice. The loss of voice frame can result in the quality degradation of the lost frame as well as subsequent frames (Daniel, 2001).

Packet delay: It is defined as the time taken for information to move from the sender to the receiver (Daniel, 2001).

Packet latency/jitter: It is the measure of time between when a packet is expected to arrive to when it actually arrives. It is the random variation in the end-to-end delay (Daniel, 2001).

RESULTS AND DISCUSSION

Higher bit rate codecs used in IP telephony systems improve speech quality, but requires more bandwidth. The numbers of voice calls are large, it creates bandwidth problem and leads to congestion in the network, eventually resulting in large delay, jitter and packet loss. The performance problems will also be experienced, in this case, by all voice calls and also by other traffic (i.e., TCP traffic) sharing the best effort IP network. In this study, we briefly describe the related contributions: The TCP friendly equation rate congestion control mechanism and the cryptographic active voice transcoding for voice flows. We have coupled the above two mechanisms to develop our proposed CAVT-ER congestion control scheme. And also this study describes the proposed system architecture, CAVT-ER congestion control algorithm and performance evaluation of CAVT-ER algorithm.

Tcp friendly equation rate congestion control scheme for VOIP: The rate control is an important issue for VoIP applications using Unresponsive Transport Protocols (i.e., UDP and RTP). Many schemes were developed based on TCP-Friendly control mechanisms. These mechanisms can be classified into three main categories: Equation-based mechanisms, window-based mechanisms and Additive Increase, Multiplicative Decrease (AIMD) mechanisms. TCP Friendly Equation-based rate control is a widely popular rate control scheme over wired networks, also known as TCP-Friendly Rate Control (TFRC). In this scheme, the sender uses an equation characterizing the allowed sending rate of a TCP connection as a function of the RTT and packet loss rate and adjusts its sending rate according to those measured parameters. A key issue is than to choose a reliable characterization of TCP throughput. A formulation of the TCP response function was derived in Sze *et al.* (2002) it states that the average throughput of a TCP connection is given by:

$$T\,(\text{Bytes/sec}) = \frac{S}{t_{RTT}(\sqrt{2P/3}) + t_{RTO}(3\sqrt{3P/8})\,P(1+32P)} \quad (1)$$

Where, T-Throughput; S- Packet size; t_{RTT} - Round trip time; P-Packet loss rate; t_{RTO}-Retransmit time out (= 4 t_{RTT}). Equation 1 roughly describes TCP's sending rate as a function of the frequency of loss indication P, round trip time t_{RTT} and packet size S. This equation reflects TCP's retransmit timeout behavior, as this dominates TCP throughput at higher loss rates. In the scheme proposed in Sze *et al.* (2002) the receiver acknowledges each packet and at fixed time intervals the sender estimates the packet loss rate experienced during the previous interval and updates the sending rate using Eq. 1. This scheme updates the sending rate at fixed time intervals; hence it is suitable for use with VoIP applications. But in our study, the proposed scheme is applied at an active node and verifies the occurrence of the congestion or network overload and network under load using inequality Eq. 2 and 3, respectively as given:

$$T(\text{Bps}) \le C * \frac{S}{R * \sqrt{P}} \quad (2)$$

$$T(\text{Bps}) \ge C * \frac{S}{R * \sqrt{P}} \quad (3)$$

The inequality Eq. 2 satisfies and then we apply voice transcoding, to change the voice coding format such as high coding rate (G.711) to low coding rate (G.723.1) The inequality Eq. 3 satisfies and then we apply voice transcoding, to change the voice coding format such as low coding rate (G.723.1) to high coding rate (G.711) during real time voice communication at the intermediate nodes. The sender uses the higher bit rate codec during network under load periods and uses the lower bit rate codec during network overload or congestion periods.

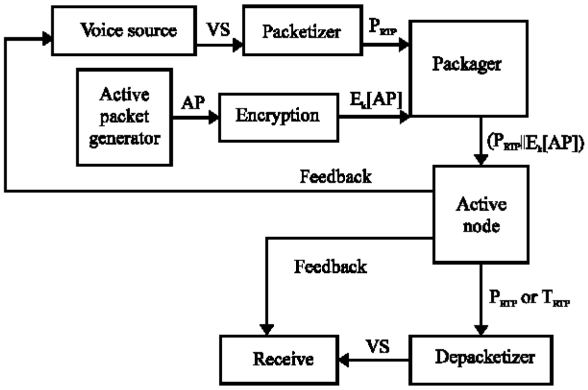

Fig. 1: Architectural block diagram

Cryptographic active voice transcoding scheme for voice flows: Transcoding aims at the reduction of overhead associated with traffic load on routers. The basic assumption for the active voice transcoding is that, at any time the codec format or codec bit rate of the voice communication can be changed dynamically according to the network congestion at the active node. But changing the voice coding format at the active or intermediate node is a security concern. So for providing security and authentication to voice transmission in real time cryptographic technique (Psounis, 1999; William, 2003) is applied on the active packets.

Architecture for cavt-er scheme: The CAVT-ER scheme architecture is shown in Fig. 1. First of all the Voice Samples (VS) are generated using voice source at sender and that is encoded with Packetizer into RTP Packets (P_{RTP}) using anyone of the audio codecs. Periodically the active code generator creates an Active Packet (AP), which consists of set of data variables and methods. Then the active packets are encrypted (E_K(AP)) using anyone of the cryptographic techniques. Then the Packager can be used to combine the RTP packet with encrypted active packet. Then the packager transmits packed information to the next intermediate or to an active node. Based on the information contained in the active packet as well as the threshold value, active node performs some extra processing i.e., voice transcoding and transmitting feedback information to both sender and receiver. Then the active node forwards either original RTP packet or transcoded RTP packet to the next node or to the receiver. Finally, the depacketizer present at receiver converts back the RTP packet into voice samples.

Algorithm modules for CAVT-ER scheme
Active packet generation

Data variables:

RTT_i=Cal_RTT (); Pi=Cal-PLoss ();
Codec_ Type=Selected_Codec;
Min_RTT=Cal_Min_RTT ();
RTO=4*RTTi; IsActivePacket=TRUE;
Bit_Rate=BitRate [Selected_Codec];
Pkt_size=PacketSize [Selected_Codec];

Methods:

Threshold_level_Checker ();

Active Packet: (AP) = (Data Variables, Methods);

Encryption (AP)

E_K [AP] = Encryption_method (AP);

Packager (P_{RTP}, E_K [AP])

Concatenate ($P_{RTP,\ Separator,}$ E_K [AP]);

int Threshold_level_Checker

T=Pkt_size/($RTT_i\sqrt{2Pi/3}$) +RTO($3\sqrt{3Pi/8}$)P($1+32P^2$)
If (Eq. 2){T_{RTP} = Trancoding(Codec_type,Codec [i+1]); Selected_Codec=Codec [i+1];}
If (Eq. 3) { T_{RTP} = Trancoding (Codec_type, Codec [i-1]); Selected_Codec= Codec [i-1];}
Else no change

Procedural steps for CAVT–ER congestion control mechanism: The steps are to be involved in the proposed CAVT-ER scheme is as follows:

- First, the sender initiates the VoIP flow transmission by creating voice samples using voice source.
- Next the voice samples are encoded into voice packets using packetizer (Initially use G.711 Codec).
- Periodically, i.e., for every 5 sec the active packet is generated with member variables and invoking methods as an executable code. Once the active packet is generated, Is Active Packet variable is set to true.
- Then the executable active packet is encrypted using cryptographic technique.
- If IsActivePacket variable is true, the encrypted active packet is combined with voice packets and then transmitted by the packager to the next intermediate nodes. Otherwise, The packager just forwards the voice pack one to the next intermediate nodes.

- If the intermediate node is an active node and IsActivePacket variable is TRUE, the packet splitter is used to separate the voice packet and encrypted executable active packet. Then the decryption is applied on the encrypted executable active packet to obtain executable active packet. Then the executable active packet is executed. Otherwise the voice packet is just forwarded to the next intermediate node or to the receiver.
- During the execution of the active packet, it invokes the Threshold_level-Checker Procedure:
- If the calculated throughput is higher than the Threshold level, the Transcoding procedure is invoked to change the codec bit rate from higher to lower.
- Else if the calculated throughput is lower than the Threshold level, the Transcoding procedure is invoked to change the codec bit rate from lower to higher.
- Otherwise there is no codec bit rate change.
- Then the active node transmits either the voice packet or the transcoded packet to the next intermediate node or to the receiver.

Performance evaluation of CAVT-ER scheme: Different codecs have different codec attributes; we have taken only 2 different codecs G.711 and G.723.1 for our discussion. The attributes of these codecs are listed in Table 1 (VOIP, Bandwidth calculature). It shows that by changing the voice packet format from high codec rate of G.711 to low codec rate of G.723.1 applying our CAVT-ER scheme during real time voice communication, it makes a lot of change in VoIP flow transmission such as the bandwidth reserved by Resource Reservation Protocol per VoIP call is varied from 80Kbps to 23Kbps and data flow bit rate in the link is reduced from 64Kbps to 5.3/6.3Kbps, which directly control congestion in the network dynamically conforming to the network conditions. And also based on these codec types, we have given the requirement of the network bandwidth

Table 1: Codec details

Codec attributes	Codec types G.711	 G.723.1
Coding method	PCM	Multirate CELP
Bandwidth (Kbps)	64	6.3,5.3
Conversion Delay (ms)	<1.00	~30.00
Bit rate (Kbps)	64	5.3/6.3
IPv4/UDP/RTPheader (bytes)	40	40
Payload (bytes)	160	20/24
IP bandwidth (Kbps)	192	15.96/16.96
Mean Opinion Score(MOS)	4.3	3.8
Bandwidth reserved by RSVP per VoIP Call(Kbps)	80	23

Table 2: VOIP bandwidth calculation (Without silence suppression)

Parameters/ Codec	G.711 (64Kbps)		G.732.1A (5.3Kbps)	
Sm. Period (ms)	10.0	20.0	10.0	20.0
PPS	100.0	50.0	100.0	50.0
Payload size	80.0	160.0	6.6	13.2
IP/UDP/RTP Bw	96.0	80.0	36.8	21.2
Ethernet BW	126.4	95.2	67.2	36.4

Table 3:VOIP bandwidth calculation (With silence suppression)

Sampling Period (ms)	Without silence suppression	With silence suppression
10	7911	15822
20	10504	21008
30	11804	23608
40	12562	25125
50	13075	26150
60	13470	26941

Table 4: G.732.1-5.3KBPS max simultaneous voice calls

Parameters/ codec	G.711 (64Kbps)		G.732.1A (5.3Kbps)	
Sm.Period (ms)	10.0	20.0	10.0	20.0
PPS	100.0	50.0	100.0	50.0
Payload size	80.0	160.0	6.6	13.2
IP/UDP/RTP Bw	48.0	40.0	18.4	10.6
Ethernet BW	63.2	47.6	33.6	18.2

Table 5: G.732.1-5.3KBPS max simultaneous voice calls

Sampling period (ms)	With silence suppression	Without silence suppression
10	14880	29761
20	27472	54945
30	38699	77399
40	48076	96153
50	56306	112612
60	64362	128724

(Ethernet bandwidth) with its corresponding sampling periods, maximum simultaneous voice calls for VoIP flow transmission with and without silence suppression in Table 2-5.

CONCLUSION

By analyzing the bandwidth required and maximum simultaneous voice calls supported by the codecs G.711 and G.723.1 for a VoIP flow transmission, we have shown the amount of required VoIP bandwidth for each codec with and without silent suppression in Fig. 2 and 3, respectively. From these figures it will be clear that high bit rate codec G.711 needs approximately double the amount of VoIP Bandwidth. So after implementing the procedure CAVT-ER at an active node under network heavy load period (congestion period), from the tables and the figures that it will be implied voice packet flow across the IP network may be reduced and it would directly control network traffic and congestion in the network and indirectly lead to fast bandwidth adaptation and minimize packet loss, packet delay and jitter. In our

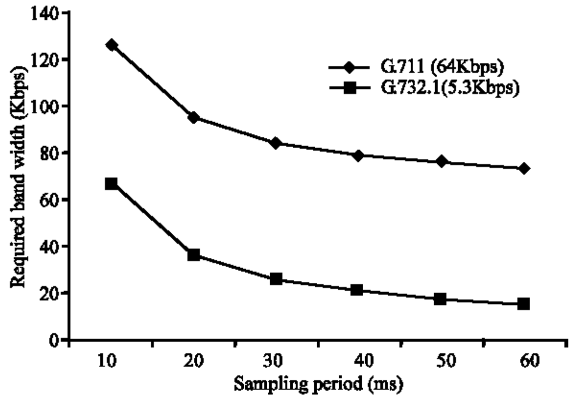

Fig. 2: Required VoIP bandwidth for codecs (Without silence suppression)

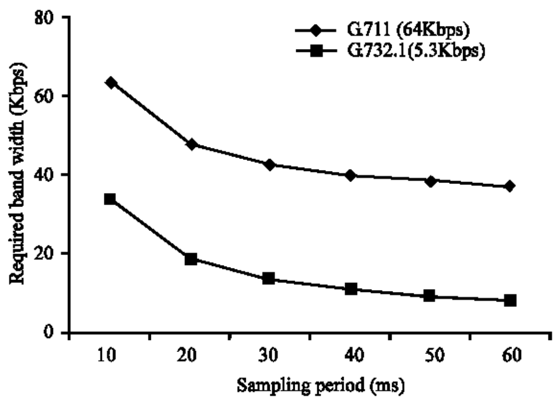

Fig. 3: Required VoIP bandwidth for codecs (With silence suppression)

study, we have analyzed how the ANS are suited to reduce the impact of congestion in an IP network with out proposed scheme theoretically. But in practical it affects many factors in the network such as QOS parameters, bandwidth adaptation, resource reservation, CPU utilization etc. In our future work, we will analyze all the above mentioned factors one by one and implement our concept in real time and analyze the tradeoff between already existing end to end congestion control schemes and our proposed scheme.

ACKNOWLEDGEMENT

I would like to thank Dr.A.M.Natarajan, my guide and his continuous support to bring out this study in a complete form.

REFERENCES

Bernet, Y., R. Yavatkar, F. Baker, L. Zhang, M. Speer, R. Braden, B. Davie, J. Wroclawski and E. Felstaine, 2000. RFC 2998, A Framework for Integrated Services Operation over Diffserv Networks.

Braden, R., D. Clark, S. Shenker, 1994. Integrated Services in the Internet Architecture: An Overview, RFC 1633.

Daniel Collins, 2001. Carrier Grade Voice Over IP, Mc-Graw Hill, Telecommunication Series.

Detti, A. M. Listanti, S. Salsano and L. Veltri, 1999. Supporting RSVP in a Differentiated Service Domain: An Architectural Framework and a Scalability Analysis, ICC.

Psounis, K., 1999. Active Networks: Applications, Security, Safety and Architectures, IEEE. Communication Surveys, Vol. 2.

Sze, H.P., Soung C. Liew, Senior Member IEEE, Jack Y.B. Lee and Danny C.S. Yip, 2002. A Multiplexing Scheme for H.323 Voice-Over-IP Applications IEEE J. Selected Areas in Commun., Vol. 20.

Tansupasiri, T., K. Kanchanasut, C. Barakat and P. Jacquet. Using Active Networks Technology for Dynamic QoS.

Tansupasiri, T. and K. Kanchanasut, 2003. Dynamic Quality of Service on IP Networks, In: Proc. of ICOIN, LNCS 2662, Springer-Verlag, pp: 573-582.

Tennenhouse, D.L., J.M. Smith, W.D. Sincoskie, D.J. Wetherall and G.J. Minden, 1997. A Survey of Active Network Research, IEEE. Commun. Mag., 35: 80-86.

VOIP Bandwidth calculator, http://www.bandcalc. com/.

William Stallings, 2003. Cryptography and Network Security, PHI.

Yan Bai and Mabo Robert Ito, 2004. University Of British Columbia, Qos Control for video and audio communication in conventional and Active Networks: Approaches and Comparison, IEEE. Communication Surveys, First Quarter, Vol. 6.