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Adaptive Rate Control - Cryptographic Active Voice Transcoding (ARC-CAVT) based Congestion Control Technique for Voice over IP (VoIP) Communication

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Abstract - **The quality of bulky real time voice communication over an IP network degrades due to packet header overhead, network congestion and improper queuing strategies. These overheads affect the overall IP network performance and Qos parameters of voice communication such as packet loss, packet delay and packet latency or jitter. The IP Network achieves best effort service by using Integrated Services and Differentiated services. In this paper, we discuss how the active networking services are suitable to reduce the impact of congestion and provide on demand Qos service in an IP network. We present system architecture for the flow of voice packets in active networking services and propose an algorithm Adaptive Rate Control-Cryptographic Active Voice Transcoding (ARC-CAVT) for Congestion Control. During congestion periods our ARC-CAVT algorithm to be invoked using active packets at an active node in a secure manner, which automatically adjusts the network throughput and Qos parameters of voice communication. Then we analyze how the ARC-CAVT Congestion Control technique should adapts the network 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. Finally, we discuss the analytical results of the ARC-CAVT performance and show that our proposed ARC-CAVT algorithm would directly control congestion in the network and indirectly lead to fast bandwidth adaptation and minimize packet loss, packet delay and jitter.**

Index Terms - **VoIP, Quality of service, packet loss, packet delay, jitter, Active Networks, Transcoding, Cryptography, Congestion**

1.INTRODUCTION

The flexibility of the Internet architecture has expedited the convergence of data communications (packet switched) and voice-based communications (traditionally circuit switched) into single IP-based core architecture. This convergence 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 have 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 header overhead, network congestion, improper queuing and hence proposes improvements.

Already existing widely used IP Network services are Integrated Services and Differentiated services achieve best effort service but not provide on demand Qos service in an IP network [1][2][3][4]. So, we develop a new architecture based on Active Networking Services (ANS) where the network participates in controlling its traffic rate and bandwidth utilization and provides on demand Qos in order to ensure acceptable performance for real time voice communication. And also, we introduce a new congestion control scheme for RTP voice traffic called ARC-CAVT, which is to be applied at an active node or at active routers during congestion or network overload periods. It adjusts the bandwidth consumption of voice packets transmitted between two end systems to reduce an impact of congestion on the network and it suddenly reduces the network traffic. ANS 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 [5].

The Active Networking Services can outperform end-to-end solutions in non-active networks for the following reasons,

- 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.
- 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.
- Speeds up the deployment of new services.
- Compatibility with existing network protocols,
- Support for incremental implementation,
- Compatibility with existing routers and network infrastructure

In [6], Tansupasiri and Kanchanasut proposed the concept of using active networks to provide dynamic QoS based on user request for an interruption of a privileged flow transmission. In [7], D-QoS system automatically adjusts its QoS settings according to user requests. In this paper, ARC-CAVT technique is to be applied at any intermediate or active node which automatically adjusts the congestion level and reduces the network traffic in the IP network without getting the user requests, improves the QOS parameters of voice communication and leads to fast bandwidth adaptation.

2. PROPOSED SYSTEM

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. Our technique ARC-CAVT is different from above mentioned mechanisms and provide effective congestion control in secure manner with less computation process. This method is similar to [8] where the adaptive rate control technique is used in integrated services environment. The ARC-CAVT architecture is shown in figure 1.

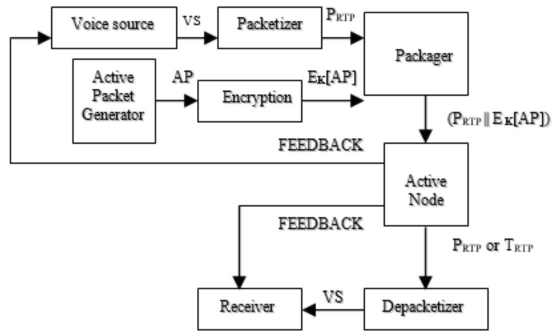

Figure 1: Architectural block diagram

2.1 Working Principle

First of all, the voice samples (VS) are generated using voice source at sender and that is encoded with Packetizer into RTP packets (PRTP) 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 (EK(AP)) using anyone of the cryptographic techniques with secret key K. The secret key K value is shared between sender and active nodes. Then the Packager can be used to combine the generated RTP packet with encrypted active packet. Then the packager transmits packed information to the next intermediate or to an active node. Now the active node splits the executable active packet from the RTP packet and executes the active code. Based on the data value contained in the active packet as well as the result of threshold level checker value, active node performs some extra processing i.e., either lower voice transcoding or higher 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 the RTP packet back into voice samples.

2.2 ARC–CAVT Congestion Control Technique for VoIP

In this technique, the network congestion rate is analyzed using adaptive rate control procedure [8]. First the procedure calculates expected amount of voice data to be transferred (1) using equation (1) and the actual amount of voice data is transferred using equation (2). Then the deviation in the amount of voice data transferred is calculated using equation (3). Then the ARC procedure uses two threshold values such as the lower threshold value (α) and the higher threshold value (β) to find out network overload period and network under load period respectively. These are calculated using equations (4) and (5) respectively.

Expected no. of voice data (EVD)=Current_Datarate x min_RTT (1)

Actual no. of voice data (AVD)= Current_Datarate x current_RTT (2)

Deviation in voice data(DVD)= AVD – EVD
= Current_Datarate(Current_RTT – min_RTT) (3)

α (bytes)= 30(Current_RTT (ms)/20(ms)) (4)

β (bytes)= 50(Current_RTT (ms)/20(ms)) (5)

Then based on the α and β values, the CAVT algorithm is applied at an active node. If DVD value is greater than β, invoke lower voice transcoding (LVT), to change the voice coding format from higher coding rate (G.711) to lower coding rate (G.723.1). If DVD value is lesser than α, invoke higher voice transcoding (HVT), to change the voice coding format from lower coding rate (G.723.1) to higher coding rate (G.711).

2.3 algorithm modules in ARC-cavt architecture

(i)Packetizer Module

Step 1: Generates RTP packet using anyone of the Codec standards such as G.711, G.723.1, G.726, G.728 etc.,

(ii) Active Packet Generation Module

Step 1: Initialize/updates the data variables as follows below,
min_RTT=*Cal_min_RTT();*current_RTT=Cal_current_RTT()*;Current_*Codec=Codec[Codec_num];Current_Datarate=DataRate[Current_Codec];IsActivePacket=TRUE;

Step 2: Create the abstract m*ethod ARC_CAVT ();*

Step 3: Generate executable active code or packet (AP), which is the bundle of above-mentioned data variables and methods.

(iii) Encryption Module

Step 1: Encrypt the executable active packet (AP) with symmetric secret key ‘K’.
E_K [AP] = Encrypt (AP);

(iv) Packager Module

Step 1: Concatenate the generated RTP packet with encrypted executable active packet with one separator ($P_{RTP,}$ Separator, E_K [AP]);

(v) ARC_CAVT procedure in Active Node

Step 1:Calculates the values for the equations (1)to (5).
Step 2: Find If DVD > β perform LVT;
Else If DVD < α perform HVT**;** otherwise does nothing.

3 SYSTEM OPERATION

The operations to be involved in the proposed ARC-CAVT architecture is as follows,

1. First, the sender initiates the VoIP flow transmission by creating voice samples using voice source.
2. Next the voice samples are encoded into voice packets using packetizer module (Initially use G.711 Codec).
3. Periodically, i.e., for every 5 seconds the active packet is generated with member variables and invoking methods as an executable code or packet. Once the active packet is generated, IsActivePacket variable is set to TRUE.
4. Then the executable active packet is encrypted using symmetric encryption.
5. If IsActivePacket variable is TRUE, the encrypted active packet is combined with generated RTP voice packets and then transmitted by the Packager to the next intermediate node. Otherwise, The Packager just forwards the voice packet alone to the next intermediate node.
6. If the intermediate node is an active node and IsActivePacket variable is TRUE, the packet splitter is used to separate the RTP 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.

7. During the execution of the active packet, it invokes the AVT_CAVT Procedure.
   a. If the DVD is higher than β , LVT is applied on the RTP voice packets. Else if the DVD is lower than α, HVT is applied on the RTP voice packets.
   b. Otherwise, there is no codec bit rate change.
8. Then the active node transmits either the RTP voice packet or the transcoded RTP voice packet to the next intermediate node or to the receiver.

4. ANALYSIS

Different codecs have different codec attributes; we have taken only two different codecs G.711 and G.723.1 for our discussion. The attributes of these codecs are listed in Table I [13][15]. 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 ARC_CAVT 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 [14] 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 (Ethernet bandwidth) with its corresponding sampling periods, maximum simultaneous voice calls for VoIP flow transmission with and without silence suppression in Table II through Table V.

5. CONCLUSION

Our proposed technique ARC-CAVT, during congestion periods to be invoked using active packets at an active node in a secure manner, which automatically adjusts the network throughput and Qos parameters of voice communication. Then we analyzed that ARC-CAVT Congestion Control technique should adapt the network 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. Finally, we concluded that the analytical results of the ARC-CAVT performance showed that it would directly control congestion in the network and indirectly lead to fast bandwidth adaptation and minimize packet loss, packet delay and jitter.

TABLE I: 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 II : VOIP BANDWIDTH CALCULATION

Parameters/ Codec	G.711 (64Kbps)		G.723.1A (5.3Kbps)	
Sm.Period (ms)	10	20	10	20
PPS	100	50	100	50
Payload size	80	160	6.6	13.2
IP/UDP/RTP Bw	96	80	36.8	21.2
Ethernet BW	126.4	95.2	67.2	36.4

TABLE III: VOIP BANDWIDTH CALCULATION (WITH SILENCE SUPPRESSION)

Parameters/Codec	G.711 (64Kbps)		G.723.1A (5.3Kbps)	
Sm.Period (ms)	10	20	10	20
PPS	100	50	100	50
Payload size	80	160	6.6	13.2
IP/UDP/RTP Bw	48	40	18.4	10.6
Ethernet BW	63.2	47.6	33.6	18.2

TABLE IV: G.723.1-5.3KBPS MAXIMUM SIMULTANEOUS VOICE CALLS

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 V: G.723.1-6.3KBPS MAXIMUM SIMULTANEOUS VOICE CALLS

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

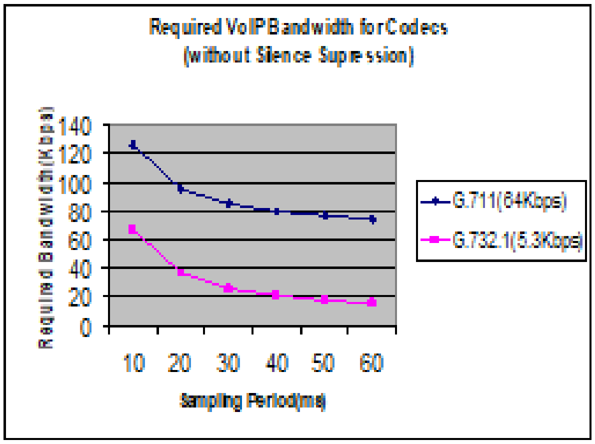

Figure 2: Required VoIP bandwidth for Codecs (Without silence suppression)

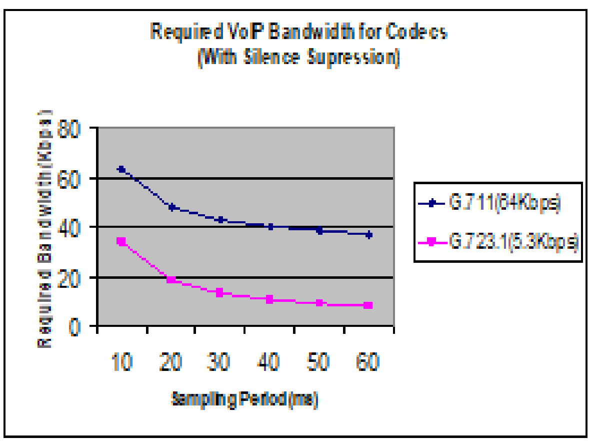

Figure 3: Required VoIP bandwidth for Codecs (With Silence Suppression)

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