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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
Real-time voice communication over IP networks faces several challenges, such as the overhead from packet headers, network congestion, and poor queuing practices. These issues affect the quality of voice communication by increasing packet loss, delay, latency, and jitter. While IP networks typically offer best-effort service, Integrated Services and Differentiated Services provide some level of quality management.

This study looks into how active networking services can help address these problems and provide on-demand quality of service (QoS). We propose a new system architecture designed to manage the flow of voice packets in an active networking environment. At the core of this system is the Adaptive Rate Control-Cryptographic Active Voice Transcoding (ARC-CAVT) algorithm, which is specifically designed to handle congestion.

When congestion occurs, the ARC-CAVT algorithm activates through special packets sent to a secure node within the network. This algorithm automatically adjusts network throughput and other QoS parameters to maintain high-quality voice communication. We explore how ARC-CAVT dynamically adapts network throughput based on the generation of voice packets during congestion, ensuring stable voice data flow for VoIP applications.

Our analysis shows that the ARC-CAVT algorithm effectively manages network congestion, quickly adjusts bandwidth as needed, and reduces packet loss, delay, and jitter, thereby improving the overall quality of voice communication.

INTRODUCTION
The adaptability of the Internet has accelerated the merging of data and voice communications into a unified IP-based system, offering a major opportunity to lower communication costs on a large scale. VoIP (Voice over IP) utilizes sophisticated audio codecs that compress voice data to very low bit rates (below 10 Kbps), using bandwidth only when transmitting voice packets. However, because voice and audio are sensitive to delays, any extended latency can significantly affect performance. To keep voice delays minimal, we can either limit the network load or implement Quality of Service (QoS) measures.

Several issues, such as packet header overhead, network congestion, and poor queuing practices, lead to inefficient use of network resources and degrade voice communication quality. Existing IP network services like Integrated Services and Differentiated Services offer only basic, best-effort service without providing the flexible, on-demand QoS required for real-time communication.

To address these limitations, a new architecture based on Active Networking Services (ANS) has been developed. ANS enables the network to control traffic rates and bandwidth utilization dynamically, ensuring better QoS for real-time voice communication. Additionally, a novel congestion control scheme called ARC-CAVT (Adaptive Rate Control-Cryptographic Active Voice Transcoding) has been introduced. This scheme manages bandwidth usage during times of congestion, minimizing the negative effects of network overload on voice packet transmission.

Active Networking Services offer several advantages over traditional end-to-end solutions:

- **Packet-Centric Approach**: Packets can carry custom code that runs at intermediate nodes as they travel through the network.
- **Traffic Reduction**: Unlike traditional methods that apply traffic control uniformly from the network's entry point to its edge, active networks apply traffic control selectively at active routers.
- **Faster Service Deployment**: New services can be implemented more quickly.
- **Compatibility**: ANS is compatible with existing network protocols, routers, and infrastructure, and supports incremental implementation.

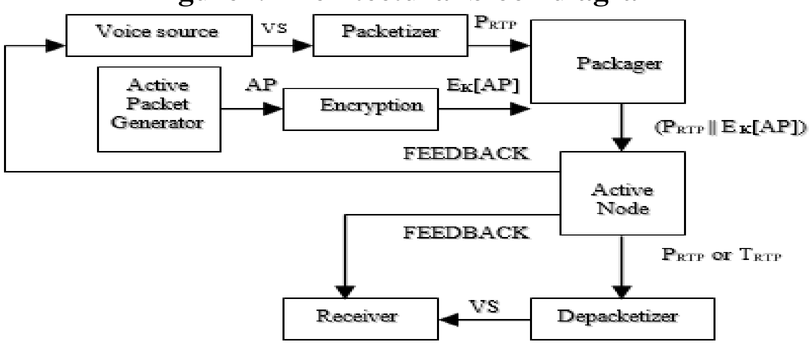
Previous research, such as the work by Tansupasiri and Kanchanasut, explored using active networks for dynamic QoS management in response to user demands. The D-QoS system also adapts QoS settings automatically based on user needs. The ARC-CAVT technique, discussed in this study, can be deployed at any intermediate or active node in the network. It adjusts congestion levels and reduces traffic without user intervention, improving the QoS for voice communication and enabling rapid bandwidth adjustments.

1. PROPOSEDSYSTEM

Our ARC-CAVT technique stands out from existing methods by offering a unique and effective solution to a critical issue in VoIP (Voice over IP) applications, particularly those using unresponsive transport protocols like UDP and RTP. While many approaches rely on TCP-friendly control mechanisms—such as equation-based, window-based, and AIMD (Additive Increase, Multiplicative Decrease) strategies—ARC-CAVT focuses on secure and efficient rate control.

Unlike traditional methods, which can be computationally intensive, our secure congestion control approach reduces the complexity of processing while maintaining robust data security. It shares similarities with the adaptive rate control approach mentioned in reference [8], which operates within an integrated services framework, but offers additional enhancements tailored to the needs of VoIP applications. The ARC-CAVT architecture is shown in figure 1.

Figure1: Architectural block diagram



2.1 Operational Principle

Here's how the ARC-CAVT technique works:

1. **Voice Sample Generation and Encoding**: Voice samples are first captured from a voice source at the sender's end. These samples are then encoded into RTP packets using any standard audio codec.
2. **Active Packet Creation**: Periodically, an active packet is generated containing specific data and methods. This active packet is then encrypted using a shared secret key known only to the sender and active nodes.
3. **Packet Packaging**: The RTP packet and the encrypted active packet are combined into a single package for transmission.
4. **Processing at Active Nodes**: When the package reaches an active node, it separates the active packet from the RTP packet and executes the active code. Depending on the data in the active packet and the results from a threshold checker, the node may perform additional processing. This could involve changing the voice codec to adjust quality based on network conditions, and then sending feedback to both the sender and receiver. The node then forwards either the original or modified RTP packet to the next destination.
5. **Receiving and Depacketizing**: At the receiver's end, the RTP packet is converted back into voice samples.

2.2 Congestion Management

The network's congestion rate is managed through an adaptive rate control process. Here's a summary of the process:

1. **Calculating Data Transfer**:
   - **Expected Voice Data (EVD)**: Computed by multiplying the current data rate by the minimum round-trip time (RTT).
   - **Actual Voice Data (AVD)**: Calculated by multiplying the current data rate by the current RTT.
   - **Deviation (DVD)**: The difference between EVD and AVD, which represents how much the data transfer deviates from expectations.
2. **Threshold Evaluation**:
   - **Lower Threshold (α)** and **Higher Threshold (β)** are determined to assess network load.
   - If the deviation (DVD) exceeds β, lower voice transcoding is applied (e.g., changing from a high-rate codec like G.711 to a lower rate codec like G.723.1).
   - If the deviation is below α, higher voice transcoding is applied (e.g., changing from G.723.1 to G.711).

Algorithm Modules in ARC-CAVT Architecture

1. **Packetizer Module**:
   - Generates RTP packets using codecs such as G.711 or G.723.1.
2. **Active Packet Generation Module**:
   - Initializes and updates data variables like minimum RTT and current RTT.
   - Creates the active packet with the necessary executable code and methods.
3. **Encryption Module**:
   - Encrypts the active packet using a symmetric secret key.
4. **Packager Module**:

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- Combines the RTP packet with the encrypted active packet and transmits them.

5. **ARC_CAVT Procedure in Active Node**:

- Calculates values needed to determine network load.
- Applies lower or higher voice transcoding based on the deviation, or does nothing if the deviation is within acceptable limits.

3. System Operation

1. **Start of VoIP Transmission**: The sender begins by generating voice samples and encoding them into RTP packets using a codec like G.711.
2. **Active Packet Handling**:

- Every 5 seconds, an active packet is created and encrypted if needed.
- The Packager combines this packet with the RTP packet for transmission.

3. **Processing by Intermediate Nodes**:

- Active nodes separate, decrypt, and execute the active packet.
- Based on the active packet's instructions, the node may apply voice transcoding and forward the processed RTP packet.

4. **Final Delivery**: The RTP packet, whether transcoded or not, is sent to the next node or to the receiver.

4. Analysis

The impact of changing from a high-rate codec like G.711 to a lower-rate codec like G.723.1 is significant. This change affects bandwidth usage and data flow rates, helping to control network congestion dynamically. For instance, bandwidth per VoIP call can drop from 80 Kbps to 23 Kbps, and link data rates can decrease from 64 Kbps to between 5.3 Kbps and 6.3 Kbps. Tables provided in the analysis detail the network bandwidth requirements and sampling periods for different scenarios, including those with and without silence suppression.

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

it was determined that the analytical findings of the ARC-CAVT performance indicate its ability to effectively manage network congestion, facilitate rapid bandwidth adjustment, and reduce packet loss, delay, and jitter, particularly in applications requiring consistent sending rates.

TABLE I: CODEC DETAILS

Codec Attributes	Code c Types	
	G.711	G.723.1
Coding method	PCM	Multi rate CELP
Band width (Kbps)	64	6.3,5.3
Conversion Delay(ms)	<1.00	~30.00
Bitrate (Kbps)	64	5.3/6.3
Ipv4/UDP/RTP header(bytes)	40	40
Payload(bytes)	160	20/24
IP band width (Kbps)	192	15.96/16.96
Mean Opinion Score (MOS)	4.3	3.8
Band width reserved by RSVP Per VoIP Call (Kbps)	80	23

TABLE II: VOIP BAND WIDTH 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/RTPB w	96	80	36.8	21.2
Ethernet BW	126.4	95.2	67.2	36.4

TABLE III: VOIP BAND WIDTH CALCULATION (WITH SILENCE SUPPRESSION)

5.Conclusion

Our suggested method ARC-CAVT is designed to be activated during congested periods by utilizing active packets at a secure active node, ensuring the automatic adjustment of network throughput and QoS parameters for voice communication. Subsequently, our analysis indicates that the ARC-CAVT Congestion Control technique must modify the network through put produced by voice packets when faced with congestion, all the while upholding a consistent voice data throughput to effectively achieve VoIP QoS.

TABLE IV: G.723.1-5.3KBPSMAXIMUM SIMULTANEOUS VOICE CALLS

Sampling Period(ms)	WithoutSilence 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

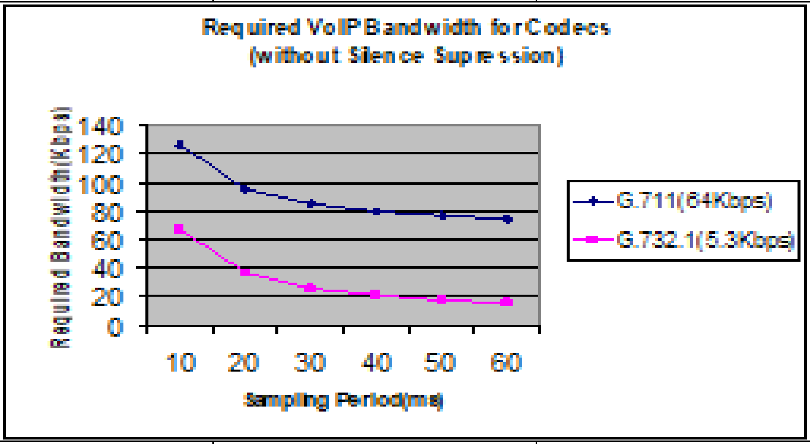

Figure2: Required VoIP band width for Codecs (Without silence suppression)

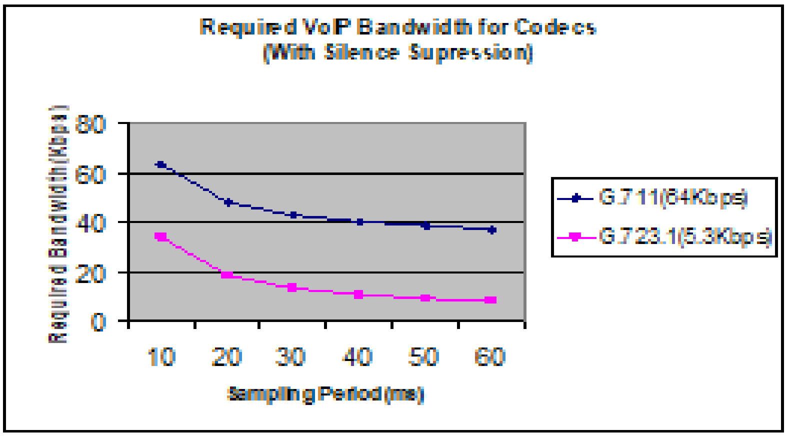

Figure3: Required VoIP band width for Codecs (With Silence Suppression)

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