NAG3

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CHAPTER 3

APPROACH & METHODOLOGY

1. ALGORITHMIC APPROACH

The encryption of audio signals is a task that requires the utmost optimisation in the user and sender's end. Keeping this in mind we did a survey of many literature papers and articles and landed on the conclusion that in order to maximize the efficiency and minimize the system load on which the program will run it is very important that we find the best suitable algorithm to solve the problem. Therefore after reading about 20+ papers on the matter subject we handpicked 6 such papers that guided us towards a solution/approach we needed.

These papers were titled as mentioned in the following order:

A. An Effective Watermarking Method Based on Energy Averaging in Audio Signals
B. Digital Watermarks for Audio Signals
C. Efficient watermarking algorithm for digital audio/speech signal
D. Digital audio watermarking for QoS assessment of MP3 music signals
E. A blind quantum audio watermarking based on quantum discrete cosine transform
F. Development of an audio watermarking with decentralization of the watermarks

The first paper proposes a spread time echo method by using pseudo noise sequencing for digital watermarking and uses the phenomenon of repetition of sound on reflection from an obstacle and encrypts it using PRN, also known as pseudo random noise. This method uses a signal on the same length as a noise and satisfies it using statistical randomness. Therefore since this algorithm uses statistical randomness, it requires the software to run many patterns and regularities that are non recognizable and therefore rely on the mean of the statistical value. We therefore came to the conclusion that this will use much of the system resources and is hence not variable for low end users.

The second paper elaborates the term "pirate" and how PN sequencing can easily decrypt normal encryption and therefore why it is important to do digital - watermarking. Ignoring the motivation of the paper and focusing solely on the method it discusses we find that the method of focus is PN sequencing. PN sequencing stands for Pseudo-random Noise sequence and uses sets of bits that are statistically meant to be random. This Approach is an improvement on the previous approach as it depends on a function that can be utilized by any software and can run even on low end systems.

The third paper discusses A blind and audio/speech watermarking algorithm that combines the discrete Tchebichef moment transform (DTMT), the chaotic system of the mixed linear–nonlinear coupled map lattices (MLNCML), and discrete wavelet transform (DWT). In addition, the adopted strategy has a blind nature, where no original audio/speech is needed in watermark extraction. This paper came from M. Yamini, H.Karmouni and M.Sayyouri and uses a very well optimized algorithm that not only improves the process of audio encryption but at the same time provides Tchebichef moment transform (DTMT) method to compress the digital images. It uses a novel set of orthogonal moments applied in the fields of image analysis and pattern recognition.

Although this algorithm is extremely fast and provides a solution for the image compression as well, we voted to not use this as the scope of mathematics required to implement the method is way beyond our scope and will not meet/line up with the deadline of the project.

The fourth paper discusses Digital audio watermarking for QoS assessment of MP3 music signals. This is another paper that discusses the type of audio signal that we are aiming at. And at the same time this paper proposes an audio watermarking signal processing technique to provide a quality assessment of the received audio signal after coding/transmission process. But this paper has no mention of any encryption algorithm but uses signal processing to assess the quality of the signal. But since this was the only paper (among few) that talks about the quality assessment of audio signal using an open source software we found this approach to be quite helpful for using in the testing phase of our project.

The fifth paper discusses A blind quantum audio watermarking method based on quantum discrete cosine transform. It uses the integration of LSB (least significant bit) and MSB (most significant bit) based on quantum discrete cosine transform (qDCT). The quantum discrete cosine transform expresses a finite sequence of data points in terms of a sum of cosine functions oscillating at different frequencies. This is a very advanced approach as it requires a very large dataset to fine tune the regression value. It also requires some capital to get the dataset and the software required to work on. But since our work is focused on an open source approach we decided to not go with this approach.

The sixth paper talks about the Development of an audio watermarking with decentralization of the watermarks. The proposed technique is executed by incorporating multi-level DWT along with the use of multiple images with different sizes as watermarks. This approach uses Discrete Wave Transform. In general sense a Discrete Wave Transform always returns only a single coefficient to its approximation value. Using the same on a multi level signal allows the extraction of multilevel time-frequency features from a time series by decom- posing the series as low and high frequency sub-series level by level. This is also a wonderful approach that does not over complicate the signal transposing value. However using discrete signal analysis will require additional hardware components (ADC - DAC Filters) to be integrated and this will in result require each sender and receiver encrypting their audio to buy/install a hardware component. Therefore it would increase the cost of encryption which will limit the benefit to only a handful.

Therefore after analyzing the approach discussed in the above mentioned 6 papers we have come up with the decision to use the method discussed in the 4th paper (QoS assessment) and use the quality assessment of signal processing and how to improve on it from the second paper (PN sequencing).

We found that using PN sequencing to test the encryption with digital watermarking proves to be the most fruitful as pseudo-noise code (PN code) or pseudo-random-noise code (PRN code) has a spectrum similar to the random sequence of bits that the random function provides. Some of the most commonly used sequences in direct-sequence spread spectrum systems are maximal length sequences, Gold codes, Kasami codes, and Barker codes. These are also the same max length sequences that are followed as a protocol by most transmission services so it becomes the perfect approach to carry out the project.

2. ABIDING SERVICES

Before using the QoS assessment encryption it is important to understand the QoS service.

The QoS is known as the quality of services and refers to any technology that deals with data transmission. It is in place to reduce the packet loss, jitters and latency on the network. There are priorities in place that every transmission must follow to abide by these protocols. As applications such as phone, video, and time-sensitive data transit a network, enterprise networks must deliver predictable and quantifiable services.

QoS is utilized by organizations to fulfill the traffic requirements of sensitive applications like real-time voice and video, as well as to minimize quality deterioration caused by packet loss, delay, and jitter as mentioned before.

Therefore while encrypting the audio it is absolutely necessary that the class of service is strictly followed (CoS). This comprises of certain quantitative parameters such as:

a. **Packet loss** - When network links get crowded, routers and switches begin losing packets. When packets are lost during real-time communication, such as voice or video conversations, jitter and pauses in speech can occur. When a queue, or line of packets waiting to be transmitted, overflows, packets might be lost.

b. **Jitters** - This is caused by network congestion, time drift, and route modifications. Jittering can decrease the quality of voice and visual communication.

c. **Latency** - This is the amount of time it takes for a packet to travel from its origin to its destination. Latency should be kept as low as feasible. Users may encounter echo and overlapping sounds if a voice over IP call has a significant latency.

d. **Bandwidth** - This is the capacity of a network shared channel to transmit the most data from one point to another in a given length of time. QoS improves network performance by controlling bandwidth and allocating more resources to important and urgent applications with demanding needs.

e. **Mean Opinion score** - Also known as MOS. This rates the voice quality using a 5 point scale. Where 5 is the highest value and therefore represents the highest quality.

Without QoS, data transmission can become fragmented, crowding connections to the point that performance worsens or, in certain incidents, the network outright shuts down. Quality of the service is crucial because businesses must provide consistent services for their workers and consumers to use.

The service quality determines the quality of the experience (QoE). Internal and external customer relationships may be impacted if an organization's services are untrustworthy. In this case we are assuming the client to be the receiver and the service to be ourselves for the sake of the explanation.

3. ADVANTAGES

The essential advantage of QoS is that it guarantees the availability of servers and the services that depend on it. It facilitates the secure and convenient movement of data over the network. QoS also enables enterprises to make better use of their existing bandwidth rather than upgrading network equipment to increase capacity.

Some of which can be condensed into:

- **a.** Mission-critical applications have access to the resources they require.
- **b.** Administrators can manage traffic better.
- **c.** Organizations can reduce costs by eliminating the need to purchase new network infrastructure.
- **d.** User experience is improved.

Performance of service tools select packets in order to make the most of their network's limited bandwidth. In other words, the connectivity can only transfer so much data in a given length of time. As a result, QoS tools select packets in such a way that bandwidth is managed to give the best network infrastructure feasible in the time allotted.

Packets relevant to a call, for example, would take precedence over packets connected to an email file. This is due to the fact that a direct transmission that is happening live is a more contemporaneous medium of speech than a method that is then provided by an email, which again must occur in real time. When a packet is lost or interrupted during or after a teleconference, the client application may notice jitter or lag. If packets are missed or delayed during the emailing transaction, they can indeed be delivered later and the terminal will not be affected. They will not see the email until all of the packets have been composed, but someone streaming video will see the packets as they come.

To classify packets, a QoS spatial analysis analyzes the packet headers. Packet headers are chunks of data that notify the application and other communication ports whatever the register holds, where it's going (its destination's IP address), and what it'll be used for. However, for our project which uses digital watermarking of audio signals, we will not require the IP address of the receiver to be used in the code as that function is provided by the transmission services and they take care of such stuff for due to the compliance of their routing protocols.

The QoS tool may also scan the incoming packets and detect if a packet is relevant to video streaming, choosing it above less time-sensitive packets. The shipping and return ids on something like a physical package can be referred to as packet headers. To select priority, the QoS tool modifies a section of the packet header.

For example, voice traffic can be assigned a higher priority than other types of traffic. Packets are assigned priorities using Differentiated Services Code Point (DSCP) for classification. DiffServ also uses per-hop behavior to apply QoS techniques, such as queuing and prioritization, to packets.

4. IMPLEMENTATION

For this phase we divided the work into 5 basic stages and 3 sub stages that we can look up to (but will not be used) to check if the data protocol is being followed.

I. **Planning** - The watermarking should take into consideration each framework's service needs and requirements, choose a fitting model and cultivate effective encryption of the audio signal.
II. **Design** - The watermarking should take note of all significant software changes and apply the coded QoS model to the specifics of its network architecture.
III. **Testing** - The code should test QoS settings and policies in a controlled testing environment where bugs can be worked out and solved.
IV. **Deployment** - Segments should be deployed in phases. We may choose to roll out code segments by network segment or by separate QoS function.
V. **Monitoring and analysis** - Watermarking should be adjusted to improve performance according to performance data.

Then we structure the models of implementation and condense in into 4 parts:

I. **Best Effort** - A QoS approach in which all packets have the same priority and packet delivery is not guaranteed. When networks do not have QoS policies specified and even when the architecture does not support QoS. Best Effort is used.
II. **Integrated Services (IntServ)** - A QoS strategy that allocates bandwidth along a specified network path. Applications request resource reservations from the network, and access points monitor packet flow to ensure that connected devices can process the packets.
III. **Implementing IntServ** - Requires IntServ-capable routers and makes use of the Resource Reservation Protocol (RSVP) to reserve network resources. IntServ's scalability is restricted, and it consumes a lot of network resources.
IV. **Differentiated Services (DiffServ)** - A QoS approach in which network devices including routers and switches are designed to support several types of traffic with varying priority. Traffic flow must be characterized based on a group's setup.

The network design also has an impact on how an organization deploys QoS. A private link provides end-to-end QoS along a single channel in a Multi - protocol label Switching (MPLS) network. MPLS SLAs define bandwidth, QoS, latency, and uptime. An MPLS, on the other hand, might be costly for businesses.

SD-WAN employs a variety of connectivity methods, such as MPLS and broadband. SD-WAN checks the status of existing cable connections for performance concerns and leverages its statement allowing modes to fail over depend on state. For example, if packet loss on one connection exceeds a specific threshold, SD-WAN capabilities will seek another connection.

5. TOOLS AND MECHANISM

QoS mechanisms are classified according to the functions they perform in network management.

I. **Classification and marking** - These tools distinguish between apps and arrange packets into distinct traffic kinds. Marking identifies every transaction as either a component of a network class, allowing network devices to identify the packet's class. These are executed on devices like routers, switches, and access points.

II. **Congestion management** - To identify which priority to place packets in, these technologies employ packet categorization and marking. Priority queuing, first-in, first-out, and low-latency queuing are examples of congestion management tools.

III. **Congestion avoidance** - When there is congestion in the network, these programmes monitor it and discard low-priority packets. Evenly distributed irregular rapid identification as well as random early intervention are two strategies for avoiding traffic congestion.

IV. **Shaping** - These technologies alter network traffic and favor real-time applications above less time-sensitive apps like email and messaging. Buffers, Generic Traffic Shaping, and Frame-Relay Traffic Shaping are examples of traffic shaping technologies.

V. **Link efficiency** - These programmes optimize bandwidth use and minimize network packet latency. Link efficiency methods, while not just for QoS, are used in combination with other QoS strategies. Authentic Routing Algorithm, header decompression, Tcp, header compression, and link compression are examples of link efficiency technologies.

QoS tools generally fall into these categories:

I. **Classification** - Work as identification of sent traffic and simultaneously marks it to make sure that other network devices can identify and prioritize it.

II. **Queueing** - Reserves the network bandwidth (our project however will not require queuing) to sustain packets in a buffer to process later.

III. **Policing** - Enforces a specific bandwidth limit and drops packets that don't adhere to the rule. This is a very important part of congestion avoidance. This is of utmost importance as it makes the watermark less clogged.

IV. **Shaping** - Similar to policing but queues the excess traffic in a buffer instead of completely dropping them. This, along with queueing, is part of congestion management.

V. **Weighted random early discard (WRED)** - Drops low priority data points and flows to preserve high-priority data from the negative effects of network congestion.

VI. **Fragmentation and compression** - Lowers bandwidth on a network to prevent delay and jitter.

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ORIGINALITY REPORT

38%	28%	6%	26%
SIMILARITY INDEX	INTERNET SOURCES	PUBLICATIONS	STUDENT PAPERS

PRIMARY SOURCES

1	searchunifiedcommunications.techtarget.com Internet Source	13%
2	Submitted to Info Myanmar College Student Paper	6%
3	Submitted to BAC International Study Centre Student Paper	3%
4	M. Yamni, H. Karmouni, M. Sayyouri, H. Qjidaa. "Efficient watermarking algorithm for digital audio/speech signal", Digital Signal Processing, 2021 Publication	2%
5	Submitted to University Tun Hussein Onn Malaysia Student Paper	2%
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11	Mohsen Yoosefi Nejad, Mohammad Mosleh, Saeed Rasouli Heikalabad. "A blind quantum audio watermarking based on quantum discrete cosine transform", Journal of Information Security and Applications, 2020 Publication	1 %
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14	Amita Singha, Muhammad Ahsan Ullah. "Development of an Audio Watermarking with Decentralization of the Watermarks", Journal of King Saud University - Computer and Information Sciences, 2020 Publication	1 %
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