Adaptive Packet Loss Concealment with Real-Time Queue Management for Network Audio Applications

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Abstract

This paper presents a comprehensive packet loss concealment (PLC) system designed for real-time networked audio applications, featuring adaptive queue management and autoregressive prediction. The system's architecture combines multi-threaded packet buffering with Burg algorithm-based audio prediction to maintain continuous audio playback despite network packet loss. Key innovations of the "Regulator" system include an adaptive worker thread that dynamically adjusts queue target sizes based on underrun patterns, per-channel autoregressive modeling for high-quality audio prediction, and comprehensive real-time performance monitoring. The system supports uncompressed audio flows with multiple channels, variable bit depths, and different frame sizes while maintaining low-latency operation suitable for interactive audio applications. Experimental evaluation demonstrates effective concealment of packet losses up to [X]% with minimal perceptual artifacts, while the adaptive queue management reduces underruns by [Y]% compared to fixed-size buffering approaches. The architecture's modular design enables deployment across various network audio scenarios, from music performance to voice communication systems. Statistical analysis reveals consistent performance under varying network conditions, with automatic adaptation to network jitter and variable latency patterns. The implementation provides a practical solution for maintaining audio quality in unreliable network environments while preserving the real-time constraints essential for interactive audio applications.

1 Introduction

Network audio applications face significant challenges when operating over unreliable networks where packet loss, jitter, and variable latency can severely degrade audio quality[12, 15]. Traditional approaches to handling packet loss in audio streams typically involve either simple silence insertion, packet repetition or basic interpolation techniques, all of which produce audible artifacts that compromise the listening experience[6, 14]. For interactive applications such as networked music performance, voice communication, and remote audio collaboration, maintaining continuous, high-quality audio output is critical for user experience and application viability[13, 3].

The problem becomes particularly acute in real-time scenarios where buffering must be minimized to maintain interactive responsiveness[4]. Unlike streaming applications that can afford larger buffers to smooth over network irregularities, interactive audio applications require solutions that balance audio quality preservation with minimal latency penalties[1]. This necessitates sophisticated packet loss concealment strategies that can predict missing audio content with sufficient quality to mask brief interruptions.

Burg's algorithm is a maximum entropy method for autoregressive spectral estimation that has been deployed in speech coding and audio signal processing applications[2, 10]. This paper introduces a novel packet loss concealment architecture with Burg as its prediction mechanism and "Regulator" as the engine that adapts to incoming packet flows. The system is suited for use by endpoints of various kinds including peer-to-peer and hub-and-spoke (server-mediated) topologies[8]. Using the latter, uncompressed audio "rooms" in the cloud can be engineered to serve large aggregates of clients .

The name "Regulator" is loosely inspired by an analogy to regulators used in SCUBA breathing apparatus. It takes after the concept that a SCUBA diver pulls air from a flow being pushed by the tank pressure. If

the flow slackens, then the demand valve engages but if the flow exceeds demand, then pressure relief occurs. Breathing evenly depends on having extra oxygen when necessary, and in our case PLC kicks in when the normal regime of regular packet flows is interrupted.

In a large server-mediated, cloud room with potentially 100's of clients, separate instances of Regulator ride each of the real-time flows at both ends, server-side and client-side, adapting to the flow qualities. The resulting audio experience is much improved even on less well-provisioned endpoints using, for example, WIFI or LEO satellite connectivity.

2 Related Work

Packet loss concealment techniques for audio have evolved significantly over the past decades[16, 5]. Early approaches focused on simple interpolation and repetition methods[14], while more recent work has explored machine learning-based prediction[9, 7] and adaptive buffering strategies[1, 11].

3 Method

The system described in this article is a buffering strategy for incoming packet flows that was developed for JackTrip, a network music performance application. JackTrip provides user-selectable buffering strategies allowing Regulator to be compared to earlier, more traditional approaches as will be detailed below. Regulator's multi-layered design centers around four primary components that work in concert to provide seamless audio playback despite network irregularities.

1. Regulator Class
2. BurgAlgorithm Class
3. Channel Class
4. RegulatorWorker Class

The outer Regulator Class replaces the alternative ring buffer (or "jitter" buffer) classes available in JackTrip. Each shares the same insertion point in the audio processing scheme and provides a mechanism to cushion late arriving packets. Usual methods add delay and also provide some kind of glitchy packet subsitute when the buffer goes empty. Regulator is designed to pass the received stream through the system as near to "now" as possible (no added delay) and provide a way to fix up discontinuities in the signal. Packets received at the insertion point are dispatched in a way that either leaves the signal untouched or invokes the packet loss concealment (PLC) methods of the BurgAlgorithm Class.. Since PLC should be computed per channel, individual channels are instantiated as Channel Class objects. The RegulatorWorker Class provides the multi-threaded implementation needed to balance computational load.

3.1 Packet Interface

JackTrip audio packets carry uncompressed audio and JackTrip instances emit a sequence of packets at regular intervals. JackTrip instances receive incoming packets. The packets that are in flight on the network carry sequence numbers in their headers but they do not have originating time stamps. A reasonable assumption is that they were clocked out evenly, with a high degree of precision, onto the network using the internal audio timing mechanism of the transmitting instance.

3.1.1 Push / Pull

Regulator's front-end dispatcher (Regulator::pushPacket) is called directly by the receiver instance's UDP socket listener when packets hit the socket. It timestamps incoming packets with the local time on arrival. The most recent incoming sequence number received is held in the Regulator::mLastSeqNumIn variable. Packets and their timestamps are temporarily stored in arrays indexed by their sequence number.

The dispatcher back-end algorithm (Regulator::pullPacket) method gets called at regular intervals by the audio backend (Jack, RtAudio, etc.). This method can either be associated with an audio backend callback function or can be called syncronously in non-callback enviroments (polled, like Chuck). Dispatcher's output is a packet which we'll call the "output packet."

mLastSeqNumOut is the sequence number of the most recent output packet. On the next call to pullPacket mLastSeqNumOut is incremented by 1 such that it corresponds to the expected index of the next logical packet, namely what it should be if the stream were behaving regularly. Timings of packet arrivals in the vicinity of the next logical packet are then checked. Lost packets are simply ignored because they aren't present in storage and thus can't be checked.

Any packets that arrived before the next logical packet are discarded. The remainder are searched for the first packet to have arrived within a window before the time of the present call (aka, "now"). That packet is then deemed the correct output packet and mLastSeqNumOut records its index.

3.1.2 Handling discontinuities

Discontinuities occur either because there's an overrun, in which case the method will skip ahead, or because there's an underrun and no candidate packet was found. For either case, the PLC algorithm attempts to conceal the discontinuity and produce the next output packet. When skipping ahead, PLC will interpolate across the discontinuity and when filling a gap it will predict what the data would have been. The index in mLastSeqNumOut either skips ahead or remains fixed, respectively.

3.1.3 Limitations

Sometimes too much data is missing to make a prediction. These "bad" underruns can happen for a variety of reasons, for example, a client disconnecting improperly or a long network hiccup. PLC depends on recent signal history when making its prediction and if the size of the gap is beyond "normal," it fails.

When a signal stalls like this, repeated calls of the dispatcher will needlessly invoke PLC and drive up the processing load. Such flailing can lead to an unbalanced or even unresponsive system.

To avoid that, before invoking PLC, the time of arrival of the mLastSeqNumOut packet gets checked to see how long of a gap it's going to be dealing with. If it's above a threshold (for example, 30 msec) then the dispatcher continues without PLC and instead produces an output packet of zeroes (mute packet).

3.2 Core Components

3.2.1 Regulator Class

The central `Regulator` class inherits from `RingBuffer` and serves as the main orchestrator for packet loss concealment operations. It manages the interface between incoming network packets and outgoing audio callbacks, handling timing synchronization, adaptive tolerance management, and coordination between prediction and buffering subsystems. The class maintains separate frame-per-packet (FPP) configurations for local audio interface (`mLocalFPP`) and peer network packets (`mPeerFPP`), enabling operation across different buffer sizes and sample rates.

The Regulator's initialization sequence is triggered when the first peer packet arrives, at which point it determines the peer's FPP configuration and establishes the prediction window size. The system automatically calculates the number of historical packets needed (`mPacketsInThePast`) based on the peer FPP, ensuring sufficient training data for the Burg algorithm while adapting to different packet sizes. For smaller FPP values (less than 128 frames), the system increases the history window to maintain prediction quality, while larger FPP values can operate with minimal history due to the increased information content per packet.

The class implements sophisticated timing management through multiple statistical analyzers that continuously monitor both incoming packet timing and audio callback intervals. This dual-monitoring approach enables the system to adapt to asymmetric network conditions where send and receive timing patterns may differ significantly. The tolerance mechanism automatically adjusts buffer headroom based on observed jitter patterns, glitch rates, and long-term network behavior trends.

The dispatcher frontend (Regulator::pushPacket) gets called directly by the receiver's UDP socket listener when packets hit the socket. It timestamps incoming packets with local time on arrival. The most recent incoming sequence number received is held in the Regulator::mLastSeqNumIn. Packets and their timestamps are temporarily stored in arrays indexed by their sequence number.

3.2.2 BurgAlgorithm Class

The `BurgAlgorithm` class implements the maximum entropy method for autoregressive coefficient estimation and prediction, forming the core of the packet loss concealment capability. The algorithm operates in two distinct phases: training, where it analyzes historical audio data to compute optimal AR coefficients using the Burg recursion, and prediction, where these coefficients are applied to forecast missing audio samples.

The training phase employs the Burg method's forward and backward prediction error minimization to determine autoregressive coefficients that best model the spectral characteristics of the incoming audio signal. The implementation includes several numerical stability enhancements critical for real-time operation: damping factors prevent coefficient explosion during periods of low signal energy, epsilon guards protect against division by zero conditions, and the reflection coefficient computation uses enhanced precision arithmetic to maintain accuracy across varying signal conditions.

During prediction, the algorithm applies the computed coefficients to extrapolate missing audio samples based on the available history. The prediction horizon extends beyond the immediate missing packet to pre-compute future samples, enabling smooth cross-fading when real packets eventually arrive. The system maintains separate coefficient sets for each audio channel, allowing independent modeling of stereo or multi-channel audio streams where each channel may exhibit different spectral characteristics.

3.2.3 Channel Class

Each audio channel is managed by a dedicated `Channel` instance that maintains independent state for prediction operations. This per-channel architecture is essential for proper handling of stereo and multi-channel audio streams, where each channel may have distinct spectral characteristics requiring independent autoregressive modeling.

The class manages an extensive set of audio buffers to support the prediction pipeline: `realNowPacket` stores the current received audio data, `predictedNowPacket` contains generated replacement audio for missing packets, `outputNowPacket` holds the final output after any cross-fading operations, and `futurePredictedPacket` pre-computes samples for smooth transitions when real packets arrive after a loss period.

A circular ring buffer (`mPacketRing`) efficiently stores recent packet history for training data, implementing a sliding window approach that automatically discards the oldest packets as new ones arrive. The ring buffer size is dynamically determined based on the peer FPP configuration, ensuring sufficient history for reliable prediction while minimizing memory usage. Additional buffers support the maintenance of prediction coefficients (`coeffs`), intermediate processing results, and cross-fade envelope calculations.

The Channel class also implements the critical cross-fading mechanism that ensures smooth audio transitions. When returning from a predicted packet sequence to real audio data, the system applies complementary fade curves that gradually transition from the predicted future samples to the newly arrived real audio. This prevents the audible discontinuities that would otherwise occur due to prediction errors or timing misalignments.

3.2.4 RegulatorWorker Class

The optional `RegulatorWorker` provides asynchronous packet processing through a dedicated high-priority thread, addressing performance constraints when prediction computations become too expensive for real-time operation in the main audio thread. The system automatically enables the worker when prediction processing time exceeds 70% of the local audio callback interval, ensuring that audio delivery remains uninterrupted regardless of computational load.

The worker operates using a sophisticated queue management system built around the `WaitFreeFrameBuffer` lock-free data structure. This design eliminates the possibility of priority inversion or thread blocking that could compromise real-time audio delivery. The queue implements adaptive sizing that responds to underrun

conditions by incrementally increasing the target queue depth, allowing the system to automatically adjust to varying computational loads and network conditions.

Communication between the main audio thread and the worker thread uses atomic operations and carefully designed protocols to ensure data consistency without requiring traditional locking mechanisms. The worker maintains real-time scheduling priority and includes mechanisms for graceful degradation if the prediction queue becomes full, defaulting to silence rather than blocking the audio pipeline.

3.3 Adaptive Mechanisms

3.3.1 Tolerance Management

The system implements sophisticated adaptive tolerance control through the `StdDev` statistics class, which continuously analyzes timing patterns to optimize buffer management. Two separate statistical analyzers track push timing (incoming packets) and pull timing (audio callbacks), computing rolling means, standard deviations, minimum and maximum values, and long-term trends using exponentially weighted moving averages.

The adaptive algorithm adjusts `mMsecTolerance` based on observed network jitter patterns, automatically increasing headroom when glitch rates exceed configurable thresholds (0.6% by default). The system distinguishes between startup conditions, where tolerance values may be artificially high due to initialization transients, and steady-state operation, where adaptations reflect genuine network behavior changes.

The tolerance adaptation includes both immediate responses to acute network issues and gradual adjustments to long-term trend changes. Variable headroom mode allows the system to automatically discover optimal buffer depths through operational experience, while fixed headroom mode provides predictable behavior for applications with strict latency requirements. The adaptation algorithm includes safeguards to prevent excessive buffer growth and mechanisms to gradually reduce headroom when network conditions improve.

3.3.2 Packet Selection Logic

The packet retrieval mechanism in `pullPacket()` implements an intelligent selection algorithm that examines recent arrivals to find the optimal packet for playback. Rather than simply consuming packets in arrival order, the system evaluates multiple candidate packets to select the one that best balances recency, timing constraints, and audio continuity requirements.

The system maintains a circular buffer of the most recent 4096 packet slots with associated timestamps, allowing it to account for out-of-order delivery, variable latency, and burst loss patterns. When multiple packets are available, the algorithm examines each candidate packet's age relative to the current tolerance threshold, skipping packets that are either too old (indicating network delays) or represent significant temporal gaps that would cause audible artifacts.

The selection process includes sophisticated handling of edge cases such as clock drift between sender and receiver, network congestion causing delayed packet clusters, and scenarios where no suitable packets are available within the tolerance window. The algorithm maintains statistics on skipped packets and uses this information to inform the adaptive tolerance mechanism, creating a feedback loop that optimizes system behavior based on actual network performance.

3.4 Multi-Threading Architecture

The system employs a carefully designed threading model that separates time-critical audio processing from computationally intensive prediction operations. The main thread handles incoming packet reception and audio interface callbacks, maintaining strict real-time constraints essential for glitch-free audio delivery. The optional worker thread processes prediction algorithms at real-time priority, ensuring that PLC computations do not interfere with audio timing requirements.

Communication between threads uses lock-free data structures and atomic operations to minimize latency and avoid priority inversion. The `WaitFreeFrameBuffer` implementation provides bounded wait-free operations for both producers and consumers, guaranteeing that no thread will be blocked indefinitely regardless

of the actions of other threads. This is crucial for real-time audio where any blocking operation could cause audible glitches.

The worker thread activation is automatic and transparent to the application, triggered when prediction latency measurements indicate that real-time constraints are at risk. The system includes comprehensive monitoring of thread performance, queue depths, and processing times, enabling automatic tuning of queue targets and detection of performance degradation that might require worker thread intervention.

3.5 Cross-fade and Glitch Handling

To minimize audible artifacts during transitions between predicted and real audio, the system implements sophisticated cross-fading mechanisms that ensure smooth continuity. When transitioning from a predicted packet back to a received packet, the algorithm applies complementary fade curves (`mFadeUp` and `mFadeDown`) that blend the predicted future packet with the newly arrived real data.

The cross-fade implementation accounts for potential phase and amplitude differences between predicted and real audio by using pre-computed future predictions that extend beyond the immediate packet boundary. This allows the system to anticipate the characteristics of arriving real audio and adjust the cross-fade parameters accordingly. The fade curves use optimized envelope shapes that minimize perceptual artifacts while maintaining audio energy consistency across the transition.

3.6 Format and Rate Adaptation

The architecture accommodates different audio bit depths (8, 16, 24, 32-bit) through a comprehensive conversion system that maintains precision while adapting to interface requirements. The system supports mismatched frame sizes between local audio interfaces and network packet formats through the `FPPratio` mechanism, which handles cases where peer and local frame sizes differ by automatically segmenting or aggregating packets to match interface requirements.

This flexibility enables deployment across heterogeneous networks where participants may use different audio configurations while maintaining synchronized playback. The conversion system includes proper dithering for bit depth reductions, anti-aliasing for sample rate conversions, and buffer management for frame size mismatches. All conversions are performed with consideration for real-time constraints, using optimized algorithms that maintain audio quality while meeting timing requirements.

4 Methodology

[Methodology section to be filled in]

5 Experimental Results

[Results section to be filled in]

6 Conclusion

[Conclusion section to be filled in]

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