Quality based Voice Stream Multicast over Low-Power Wireless Networks

P.S.Suganya\textsuperscript{1} B.E. (M.E), M.Ganthimathi\textsuperscript{2} M.E.,

\textsuperscript{1}Student, Department of cse, Muthayammal Engineering College, Rasipuram, Namakkal Dt, Tamilnadu, India. \\
\textsuperscript{2}Professor, Department of cse, Muthayammal Engineering College, Rasipuram, Namakkal Dt, Tamilnadu, India. \\
suganyaashock@gmail.com

\textbf{Abstract}—— Internet streaming solutions are based on unicast communication and raise scalability and efficiency. Multicast communication provides a promising and viable alternative since it can vastly improve scalability and network efficiency for the aforementioned class of applications. The suitable multicast streaming solutions ready for wide-area deployment are yet to emerge. Rate-Adaptive Multicast Streaming provide mechanisms for improving multicast voice streaming over the Internet. It originates from the requirements for multicast solutions to a large number of receivers and to accommodate the latter’s heterogeneity of bandwidth capabilities. The work exploits scalable-encoded voice and utilizes layered multicast transmission on top of the Internet Protocol (IP) multicast architecture. The set of solutions includes mechanisms for server-side as well as receiver-driven rate adjustment. An algorithm that optimally stripes the scalable-encoded data into several media quality enhancing layers considering the distribution of receiver bandwidth capabilities. The underlying optimization metric is novel and incorporates transport. It provides a mapping from each receiver’s bandwidth capability onto a utility-based fairness measure. It provide to the server for discovering the bandwidth capability distribution of the active receivers, design a feedback scheme based on probabilistic polling. It allows to control the feedback traffic within statistical bounds, making the scheme flexible and scaling to very large receiver populations. The design of scalable multicast solutions is the distribution of computational tasks and the reduction of control messages. The receiver is responsible for inferring its bandwidth capability without involving the server. It adopt and improve the state-of-the-art approach for estimating the fair bandwidth share based on TCP throughput modeling. Extensive simulation results prove the applicability of the modified scheme for estimating the TCP-fair rate of a multicast receiver. It serves also for receiver-driven rate adaptation using a timer-based multicast group subscription strategy. It yields a reasonable trade-off between the user demand for smooth voice transmission and the network requirement of cooperativeness and responsiveness to congestion indication.

\textbf{Keywords}—— Rate adaptive mechanism, IP, TCP

\section*{INTRODUCTION}

\subsection*{1.1 RATE ADAPTIVE MULTICAST STREAMING}

Communication plays an important role in society and life. Radio and television as well as telecommunications and computer networks allow for the delivery and exchange of information. Computer communication networks are becoming increasingly important in providing and maintaining complex information systems that empower the society. The Internet evolved to be the platform of choice for digital communications, a trend towards service convergence can be witnessed.

It access to the Internet over cellular phones, telephony over the Internet, and Internet radio broadcast are already state-of-the-art services. The new technologies provide the end user with high-bandwidth access, computer and telecommunications networks are only insufficiently prepared for the dissemination of voice in a point-to-multipoint manner similar to that of television systems.

\subsection*{1.2 RATE ADAPTIVE MECHANISM}

Rate adaptive mechanism is for improving multicast streaming services over the Internet are investigated and designed. It identify a set of challenging issues originating from the characteristics of streaming media applications and the heterogeneity, unpredictability, and dynamics of the communication channels. It arise from the absence of global Quality of Service (QoS) in the Internet to provide soft real-time guarantees required for streaming media. It tackle the issues by developing end-to-end solutions that build on the availability of scalable codes and utilize layered transmission on top of native multicast forwarding capabilities of the Internet Protocol (IP) multicast architecture.

\section*{II. RELATED WORK}

\subsection*{2.1 VOICE OVER SENSOR NETWORK}

Wireless sensor networks have traditionally focused on low duty-cycle applications where sensor data are reported periodically in the order of seconds or even longer. To support real-time streaming of audio and/or low-rate voice even in wireless sensor networks for use in emergency situations and short term intruder detection. Wireless sensor networks are composed of low-cost battery-operated nodes which communicate across one or more hops to at least one gateway.

The limited processing power and network bandwidth available in such networks has traditionally restricted operation to applications with low duty-cycle such as infrequent sensing and monitoring, in-network data reduction and asynchronous operation. The traditional sensor network applications focuses on passive sensing and reporting, there is a growing need to support real-time streaming for voice and low-rate voice delivery in both mission-critical operations and in wide-area surveillance, particularly under emergency conditions and for in intruder detection alerts. Constraints on computing power, bandwidth, memory and energy supply in sensor networks make the problem of delivering timeliness guarantees across multiple hops especially challenging.
It presents the capacity bounds on how much real-time data a sensor network can transfer by the imposed deadlines [5]. It derives a sufficient schedulability condition for a class of fixed-priority packet scheduling algorithms and provides a theoretical basis for capacity planning. Media access schemes have been proposed for real-time communication over sensor networks, with the goal of bounding the end-to-end delay. Simulation studies show that such a scheme is effective in minimizing the deadline-miss ratios in multi-hop sensor networks.

2.2 QUALITY-AWARE VOICE STREAMING FOR WSN

QVS is particularly designed for voice transfers in short-term emergency situations. Sensor networks deployed for coal mine monitoring will suffer from changing topologies and lossy links due to complex terrain and interference from environments. QVS is designed to support on-demand and concurrent voice transfers with limited bandwidth. QVS comprises the following components that are designed to meet the above requirements:

A. EMPIRICAL VOICE MODEL

It quantifies the voice quality of a stream based on transport parameters including packet loss ratio and delay. The model is used by the source/sink of a stream for automatic voice quality evaluation and control.

B. ADAPTATION MECHANISM

It dynamically adjusts voice compression/duplication ratios to maintain desirable voice quality in face of dynamic link conditions.

C. DISTRIBUTED ADMISSION CONTROL

The algorithm assigns stream data rates based on the stream quality requirement as well as the available network capacity measured by each node locally. QVS address the problem of supporting voice streaming over wireless sensor networks (WSNs). It is to satisfy stringent audio Quality of Service (QoS) requirements despite nodes only have limited communication and processing capabilities.

It reports the design and implementation of Quality-Aware Voice Streaming (QVS) for WSNs. QVS is built upon SenEar, a new sensor hardware platform developed for high-bandwidth wireless audio communication. The primary design objective of QVS is to provide robust voice quality for concurrent voice streaming in dynamic environments. QVS employs an empirical voice model to automatically evaluate the current voice quality of streams and provide feedback for audio compression/duplication adaptation. Rate adaptive mechanism can achieve robust voice streaming in face of dynamic variation in link quality and network topology [4].

To support multiple concurrent voice stream transfers, QVS employs a distributed admission control algorithm that assigns stream data rates based on available network capacity measured by each node locally. QVS is built upon a new sensor hardware platform for high-rate audio communication. QVS consists of several novel components including automatic voice quality modeling and evaluation, dynamic voice compression and duplication adaptation, and distributed stream admission control.

2.3 FEC-BASED ERROR CONTROL FOR INTERACTIVE AUDIO

The transmission of real time audio, and especially of real time voice, over the Internet has been much in the news recently. The large number of companies sells Internet telephony software and supporting hardware Net phones, gateways, phone-like appliances, traditional voice carriers quietly investigate the matter. It is clear that the field of packet voice over the Internet has matured and that the basic building blocks are available, ranging from high quality low bit rate codec’s to standardized protocols such as RTP.

Much recent effort has been devoted to developing mechanisms to minimize the impact of loss. Rate control mechanisms attempt to minimize the number of packets lost by making sure that the rate at which audio packets are sent over a connection matches the capacity of the connection. They typically do not prevent loss altogether. An error control, or loss recovery mechanism is required if the number of lost audio packets is higher than that tolerated by the listener at the destination [2].

Typical mechanisms fall in one of two classes. Automatic Repeat Request (ARQ) mechanisms are closed-loop mechanisms based on the retransmission of the packets that were not received at the destination. Forward Error Correction (FEC) mechanisms are open-loop mechanisms based on the transmission of redundant information along with the original information so that the lost original data can be recovered from the redundant information. ARQ mechanisms are typically not acceptable for live audio applications over the Internet because they dramatically increase end to end latency.

2.4. PERFORMANCE OF VOIP IN AN 802.11 WMN

There has been a tremendous proliferation of VoIP services in both residential homes and corporate offices. The cost savings achieved by VoIP by using existing data infrastructures along with easy deployment benefits are the main reasons driving the steady growth of VoIP. VoIP over wireless LAN (WLAN) has the potential of becoming an important application due to the ubiquity of the WLAN in homes and offices. The advent of dual cell phone handset with Wi-Fi capabilities and soft phones over PDAs, carrying voice over the WLAN is gaining a significant importance.
Providing VoIP users with true mobile phone services having the freedom of roaming requires wide area wireless coverage, and IEEE 802.11-based multihop wireless mesh networks have been considered a practical solution for wide area coverage. The benefits of mesh network compared to wired LAN connecting Wi-Fi access points are:

i. Ease of deployment and expansion.
ii. Better coverage.
iii. Resilience to node failure.
iv. Reduced cost of maintenance.

A mesh network has the potential of creating an enterprise-scale or community-scale wireless backbone supporting multiple users while driving these users from using fixed phones to wireless VoIP phones.

It experimentally investigated several methods to improve the quality of VoIP over a WLAN mesh. The use of multiple interfaces, label based forwarding architecture, and packet aggregation. The methods produce considerable improvement in the operation of the mesh – with respect to capacity, QoS, or both [6].

After evaluating several design options that a label based solution is the most appropriate for carrying real-time traffic in a wireless mesh operating in the unlicensed spectrum. The architecture combines routing and label based forwarding, and addresses all aspects required to support voip over the wlan mesh: call admission, mobility, qos.

It implements a distributed packet aggregation strategy that work conserving by using mac waiting to perform aggregation, without introducing unbounded packet delays.

III.EXISTING SYSTEM

Low power Wireless Networks (LWNs) have become increasingly available for mission critical applications such as security surveillance and disaster response. It emerging low-power wireless audio platforms provide an economical solution for ad hoc communication in emergency scenarios. It develop a system called Adaptive stream Multicast(ASM) for voice communication over multihop LWNs. ASM is composed of several novel components specially designed to deliver robust voice quality for multiple sinks in dynamic environments:

i. To mitigate the impact of lossy links on voice quality, ASM employs a transport-layer feedback based Forward Error Correction (FEC) adaptation scheme to allow the source to dynamically adapt it redundancy ratio in response to the voice quality variation at sinks.

ii. ASM also uses several novel mechanisms to reduce the signaling overhead of FEC adaptation, filter the coding redundancy on a multicast tree, and account for the impact of bursty packet loss.

iii. It develops a lightweight Tree-based Opportunistic Routing (TOR) that fully exploits the broadcast opportunities on a multicast tree based on novel forwarder selection and coordination rules.

iv. ASM includes a distributed cross-layer admission control algorithm that prevents a new stream from violating the voice quality guarantees for existing streams.

ASM has been implemented on a low-power hardware platform and extensively evaluated through experiments on a test bed of 18 nodes. The experiment result shows that ASM can achieve satisfactory multicast voice quality in dynamic environments while incurring low communication overhead. It is already provided by the routing protocol because most routing metrics are generated based on the link quality estimation.

3.1 ASM

ASM has two main design objectives in mind:

i. System should assure the voice quality of multiple sinks in dynamic link quality.

ii. Network bandwidth utilization should be minimized in order to support a large number of sinks.

The design does not rely on any particular routing metric assumption as long as there exists an approach estimating the link qualities among neighbouring nodes. ASM features the following components that reside on multiple layers of network stack to ensure voice quality while minimizing the network bandwidth usage.

A. Empirical voice quality modeling.

The sink estimates application-level voice quality based on data loss rate and codec-specific characteristics. The estimation is sent back to the source and used by other components of ASM to achieve automatic online quality assurance.

B. Adaptive FEC for multicast.

Relay nodes only forward the feedback information from the sink with the highest path loss rate. The source applies an adaptive FEC scheme that can dynamically adjust coding redundancy ratio according to feedback. The relay nodes use a redundancy filter that drops over redundant voice packets received from upstream nodes before forwarding to well-connected sinks.

C. Tree-based opportunistic routing.

To improve throughput and reduce bandwidth consumption, develop a Tree-based Opportunistic Routing protocol. TOR can fully exploit the broadcast opportunities on
a multicast tree based on novel forwarder selection and coordination rules.

D. Distributed admission control.

A cross-layer admission control algorithm is designed to prevent new streams from violating the voice quality guarantees for existing streams. The algorithm can accurately estimate the available link-level network capacity in the presence of interference and also account for the transport-layer rate constraints of multiple voice streams.

3.2 COMPONENTS OF ASM

It presents the details of each component in the ASM system and how the design objectives are achieved via the interactions between the source, sinks, and relay nodes.

3.2.1. Voice Quality Modeling

Voice streaming in LWNs often suffers frequent quality variation due to network dynamics caused by unpredictable noise and interference. To maintain satisfactory voice quality, ASM employs a voice quality model to automatically estimate the current streaming quality and adjust system parameters accordingly. Mean Opinion Score (MOS) is a widely adopted subjective model that qualifies speech quality.

MOS has been shown effective in evaluating voice quality, it is subjective and hence cannot be used for automatic voice evaluation and control. The impact of delays can be effectively mitigated by using playback buffers at sinks. The experimental results show that a delay only decreases the voice quality. The quality degradation due to delay can be safely approximated as a constant as long as the delay does not exceed a certain threshold. ASM is designed to satisfy moderate voice quality requirement and hence can tolerate an end-to-end communication delay up to several hundred milliseconds that can be easily achieved on multihop wireless networks.

3.2.2. Adaptive FEC for Multicast

It introduces the background on FEC, and discusses the design of feedback-based adaptive FEC for voice streaming. Then refine the FEC scheme to address special challenges raised by multicast, followed by an extension to the FEC scheme to handle bursty packet loss.

Coding Scheme Selection

Typical FEC schemes mitigate packet loss by sending raw data packets together with additional parity packets. The receiver receives any packets in the block, it can recover the original data packets. It is less than encoded packets are received, a fraction of the data packets can be recovered.

An important performance metric of FEC is the redundancy ratio. The higher the redundancy ratio is, the more the bandwidth would be consumed. ASM employs several novel mechanisms to address the challenges. It develops a lightweight feedback scheme for adaptive stream multicast based on implicit piggyback.

The redundancy ratio at the source is adaptively adjusted based on real-time voice quality perceived by the sinks. The relay nodes on each branch of multicast tree minimize the bandwidth consumption by filtering out redundant coded packets being forwarded on branch while satisfying the quality requirement of the voice stream.

Feedback and Redundancy Filtering

To achieve automatic voice quality assurance, sinks monitor the quality of the received voice stream and send feedback to the source. It designs an adaptive scheme where each sink maintains a sliding window containing the packet loss rates during the last seconds. The sink calculates the mean packet loss rate during the last s seconds and then derives the required coding redundancy ratio.

The feedback approach explicitly takes the voice quality into consideration. To reduce the network bandwidth usage, ASM further optimizes the feedback-based FEC using two mechanisms. It sends feedback to the source using an implicit piggybacking scheme. Sinks first send the feedback to their parents on the multicast tree. Receiving the feedback, the nodes piggyback it in the header of the data packets to be forwarded.

The sink with the highest data loss rate determines the coding redundancy ratio of source, the feedback from multiple sinks is aggregated at each tree junction node. The feedback from the sink with the highest path loss rate is passed along the tree toward the source, while other smaller values are dropped during aggregation at intermediate nodes.

The redundancy ratio is adjusted according to the sink with the highest path loss rate, it may result in bandwidth waste if relay nodes forward all received packets directly to better connect sinks. The junction node on the multicast tree maintains the highest redundancy ratio required by the sinks in the subtree rooted at it. The nodes only need to maintain coding parameter.

3.2.3. Tree - Based Opportunistic Routing

Opportunistic routing has been shown to be a promising technique to improve network throughput by exploiting the broadcast nature of wireless networks. The typical OR protocols, the sender broadcasts its data among the nodes that can overhear the transmission. There exist a number of opportunistic routing schemes such as ExOR and SOAR.
1. Candidate forwarder selection. Unlike unicast, there are multiple sinks on a multicast tree. The selecting candidate forwarders by only considering the ETX from a relay node to single sink will not work. A simple adjustment is to have the sender specify several candidate lists in the packet header for different sinks.

2. Coordination between forwarders. When multiple forwarders overhear a transmission, only a subset of them should forward the packet to avoid duplicate transmissions. It is nontrivial to select the right subset and also coordinate the transmissions among them because the subset can have nodes on the paths to multiple sinks in the case of multicast.

The unique challenges described previously motivate the design of a new OR protocol called tree-based OR for stream multicast. TOR runs on top of a routing tree connecting the source to the sinks. It assumes that each link chosen by the multicast tree or TOR is bidirectional although the ETXs in two directions may be different.

Coordination between forwarders. Eligible candidates that satisfy both C1 and C2 need to be coordinated such that the most efficient candidate will actually forward the packet. An ideal coordination rule should ensure that, when candidates on a path from the source to a sink overhear the same transmission, the one closest to the sink should start the forwarding before any other candidate on the same path.

3.2.4. Distributed Admission Control

Admission control determines whether a sink can be accommodated in the network while achieving satisfactory voice quality for all admitted sinks. Existing admission control algorithms designed for ad hoc networks do not support stream multicast with voice quality guarantee and hence are not suitable for ASM. It assumes that sinks request to join the multicast tree one at a time. The assumption is reasonable because the admission control process in ASM only takes several hundred milliseconds.

The network capacity may be exceeded in two cases: when a new sink joins the network, the traffic load in one node’s contention domain may increase if the node lies on the path from the source to the new sink or it is interfered by any node on the path. The traffic load may increase if the source increases the redundancy ratio as requested by a sink. ASM predicts the traffic load of each node. The new load of any node exceeds the available bandwidth capacity, the admission of the new sink is denied.

Each sink compares the voice quality before and after redundancy ratio adjustment. The voice quality decreases with an increased redundancy ratio at the source, it implies that the network capacity may be exceeded leading to a high packet loss rate along the path. The sink will first leave the multicast tree and then request to join as a new sink. It only focuses on the case that a new sink joins the network.

Estimation of Local Capacity

ASM is that the admission control is performed efficiently in a distributed fashion. A key step of the admission control algorithm is that each node checks whether the new total traffic load would exceed the available bandwidth if a new sink were admitted.

The maximum bandwidth available at a node as local capacity. It introduces two concepts used to estimate the local capacity, contention domain and saturation rate. Contention domain of node i refer to the set of nodes who share the bandwidth.

The saturation rate is the maximum throughput observed at a receiver when all senders are within the contention domain of each other. The experiments on CC2420 CSMA MAC show that saturation rate could be a reasonable approximation of the local capacity. The saturation rate is solely dependent on the number of nodes in a contention domain and hence it can be accurately measured. It conducted experiments with SenEar nodes to measure the saturation rate under various number of nodes.

Admission Control Process

The objective of admission control is to ensure that the local capacity constraint for each node in the network should be met if the new sink is admitted. ASM, a new sink joins the multicast by sending a request with a desirable data rate to the source. It is derived by accounting for several factors including raw data rate, path loss rate, and redundancy ratio. The request is forwarded along the path and each node interfered by the new stream checks whether its local capacity constraint is violated. The new stream is accommodated if and only if the local capacity constraints of all nodes interfered by the new stream are respected.

ASM: Adaptive Voice Stream Multicast: ASM is a multilayer solution that are composed of several novel components specially designed to deliver robust voice quality for multiple sinks with minimum bandwidth usage and it use Distributed cross-layer admission control algorithm.

3.3 DRAWBACKS OF EXISTING SYSTEM

1. Require several feedback rounds before an estimate can be calculated.
2. Require time and energy consumption.

IV.PROPOSED SYSTEM

It is a developed End-to-End Mechanisms for Rate- Adaptive Multicast Streaming over the Internet.

i. It started with an assumption of global knowledge about the receiver bandwidth capabilities at the server. The data sources to optimally adjust the transmission rates, existing solutions rely on network-centric metrics.
ii. The metrics assume a linear correlation of the user satisfaction and the data rate, research in voice quality measurement has indicated a non-linear relationship between both measures.

iii. It extended the state-of-the-art inter-session fairness metric to incorporate application aspects in terms of user utility and voice quality.

iv. It developed an appropriate optimization algorithm that has reasonable low complexity to allow for frequent computation.

The proposed system use Stream Optimization Algorithm and Minimum Increase & Minimum Decrease.

### 4.1 MODULES

i. Intra-session performance metric

ii. Scalable feedback control

iii. TCP-compatible rate estimation

iv. Group subscription management

### 4.2 MODULES DESCRIPTION

#### 4.2.1. INTRA-SESSION PERFORMANCE METRIC

The intra-session performance of a multi-rate multicast session derived the receiver utility fairness function. The multicast source can map from the inferred TCP-compatible rate of a receiver to a corresponding fairness index. The state-of-the-art metrics attempt to consider network-centric aspects only, metric also enables incorporating application-specific aspects, such as the user satisfaction derived from rate-distortion curves and subjective voice assessment.

Based on the receiver utility fairness function, the intra session performance is defined as the average receiver utility fairness index of all receivers. The latter is then used to quantitatively evaluate the benefit of multi-rate multicast schemes and adaptive stream organization. And developed an algorithm for optimized organization of encoded data into a fixed-size base layer and several adjustable enhancement layers.

#### 4.2.2. SCALABLE FEEDBACK CONTROL

The developed optimization algorithm requires knowledge about the distribution of receiver’s bandwidth capabilities. Knowledge has to be gathered from the receivers, developed a feedback suppression scheme based on probabilistic sampling in order to avoid the problem of feedback implosion. It uses the sample values directly to calculate the layering scheme and quantitatively study the impact of the sample size on the performance of the optimization algorithm. It derives a statistical model for estimating the bandwidth capability distribution based on the collected samples.

To evaluate the model-based approach and compare it with first approach, a set of simulations for several theoretical and measured bandwidth distributions have been conducted.

#### 4.2.3. TCP-COMPATIBLE RATE ESTIMATION

Receivers are required to estimate their bandwidth capabilities for both choosing a reasonable subset of layers (subscription level), and feeding back the estimate to the sender for stream optimization purposes. Mechanisms utilizing a well-known equation-based approach for calculating a TCP-fair rate have been extensively evaluated for unicast streaming. The gathered results indicate that the technique provides a promising basis for smooth rate adaptation.

Only recently existing work on multicast schemes adopted the model-based approach originally designed for closed-loop unicast control. The experimental results, however, show that naively adopting the algorithm may not provide the expected behavior. Analysis and extensive simulations led to the development of an improved algorithm for equation-based estimation of the TCP-compatible rate of a multicast receiver.

#### 4.2.4. GROUP SUBSCRIPTION MANAGEMENT

Performing group subscription decisions simply based on the actual calculated TCP-fair estimate might lead to frequent join and leave decisions due to the inherent variations of the estimated value. The developed mechanism based on dynamic timers that allows for smooth subscription decisions in order to reduce oscillations otherwise caused by frequent join and leave actions.

The parameters of the mechanism are tunable such that its responsiveness to congestion indications respectively its aggressiveness regarding the allocation of available network resources can be adjusted independently. The behavior and performance of mechanism are discussed and compared to that of a very prominent existing mechanism by means of network simulations.

### V. CONCLUSION & FUTURE WORK

Adaptive Voice Stream Multicast assures the voice quality and network bandwidth utilization. It use Tree-based opportunistic routing protocol is used to improve throughput and reduce bandwidth consumption. It used distributed admission control algorithm is designed to prevent new voice streams from violating the voice quality guarantees for existing streams. But Adaptive Voice Stream Multicast requires time and energy consumption. It causes congestion and delay. So in future work Rate Adaptive Mechanism is designed in which optimization algorithm and Multiplicative Increase & Multiplicative Decrease algorithm is used to reduce complexity of voice transmissions.
REFERENCES


