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Real Time Message Broadcasting for Educational System

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ABSTRACT –

The Real-Time Audio Broadcasting System is designed for college festivals to enable instant, clear, and selective voice announcements across departments. Using audio-over-IP with RTP, IGMP, and the G.722 codec, it ensures ultra-low latency and high-quality broadcasts via multicast technology. The setup includes a microphone console, IP receivers, and a local network, allowing the principal or coordinator to communicate efficiently during events. It can also serve for emergency alerts and daily announcements. Future enhancements may include wireless connectivity, mobile app integration, and live monitoring for a more flexible and modern communication system.

Keywords: Real-Time Audio Broadcasting System, college festivals, instant, clear, selective voice announcements, departments, audio-over-IP, RTP, IGMP, G.722 codec, ultra-low latency, high-quality, broadcasts, multicast technology, microphone console, IP receivers, local network.

I. INTRODUCTION

The Real-Time message broadcasting for educational system. College festivals are one of the most vibrant and celebrated parts of student life. These events bring together cultural performances, competitions, exhibitions, and interactive sessions that involve multiple departments across the campus. While the energy and enthusiasm during a college festival are unmatched, managing such large-scale events efficiently is always a major challenge.

One of the key aspects of smooth event management is effective communication. During a festival, announcements about schedules, results, emergencies, or performance updates need to be made instantly. In traditional setups, this is usually done through handheld microphones, portable speakers, or manual runners carrying messages from one department to another. These methods are slow, inefficient, and often lead to confusion or delays.

To overcome these limitations, there is a growing need for a centralized and real-time communication system that can broadcast live audio messages to multiple departments at once — with minimum delay and maximum clarity. This is where the concept of a Real-Time Audio Broadcasting System comes into play.

II. SYSTEM OVERVIEW

The Real-Time message broadcasting for educational system designed to deliver live voice messages across multiple departments simultaneously through the college's local network infrastructure. It allows the Principal, event coordinator, or organizing committee to make instant

Announcements to specific departments or to all departments together. The system uses advanced audio- over-IP (AoIP) technology, which converts the voice input into digital data packets that are transmitted across the network using Multicast IP communication. Each department is assigned a unique multicast address, ensuring selective communication meaning an announcement can be made to one department, multiple departments, or the entire campus at once.

This approach not only saves time but also ensures that every message is delivered clearly, instantly, and without network congestion. The setup includes a paging console or sender unit, an IP PBX/SIP server, and receiver units or endpoints in each department, all connected through the college's LAN.

III. LITERATURE SURVEY

3.1 Evolution of Real-Time message Broadcasting

Real-time audio communication has evolved from traditional wired PA systems to network-based architectures that utilize Ethernet and IP protocols. The LOLA (Low Latency Audio/Video Streaming System), developed by Claudio Allchin et al. (2013) at GARR, demonstrated how ultra-low-latency audio-visual transmission could be achieved for remote musical performances. Published by Springer, the study achieved end-to-end latencies below 10 milliseconds, proving the feasibility of interactive real- time audio systems in distributed environments.

This research laid the foundation for real-time broadcasting in campuses, enabling synchronized communication across multiple departments using standard IP infrastructure.

3.2 IP Multicast and Network Efficiency

The Real-time Transport Protocol (RTP), introduced by Schulzrinne et al. (2003) in RFC 3550 (IETF), provides the framework for delivering real-time audio and video data over IP networks. RTP includes features like timestamps and sequence numbering, which are vital for maintaining timing and order in streaming applications.

Additionally, Audio/Video Bridging (AVB) and Time- Sensitive Networking (TSN) standards by the IEEE 802.1 Working Group (2012–2018) enable deterministic delivery of media data with minimal jitter. These standards guarantee that audio packets are delivered at predictable times — an essential feature for synchronized campus broadcasting. Multicast communication, managed through the Internet Group Management Protocol (IGMP), allows one-to-many data distribution, reducing network load compared to multiple unicast streams. As noted by Pannell et al. (IEEE, 2008), multicast significantly improves bandwidth efficiency and system scalability, making it ideal for campus-wide announcement systems.

3.3 Low-Latency Audio Codecs

The choice of codec determines the clarity and responsiveness of real-time audio. The Opus codec, defined by Jean-Marc Vain et al. (2012) in RFC 6716 (IETF), was Developed for low-delay, high-quality communication. Opus provides frame sizes as low as 2.5 milliseconds, achieving superior latency and adaptability over fluctuating networks.

In the field of interactive music and voice transmission, Turchet et al. (2022) presented ―Networked Music Performance—State of the Art‖ in Frontiers in Signal Processing, highlighting that maintaining total latency below 25–30 milliseconds is critical for natural real-time interaction. This aligns with the latency goals of the proposed college broadcasting system.

3.4 Synchronization and Professional Audio Standards

Synchronization ensures all speakers across the campus play the same audio simultaneously. IEEE 802.1AS (2011) and SMPTE ST 2110 (2018) standards define timing and transport mechanisms for professional broadcast media over IP networks. SMPTE (Society of Motion Picture and Television Engineers) developed ST 2110 to separate audio, video, and metadata streams, allowing precise synchronization and high-quality playback.

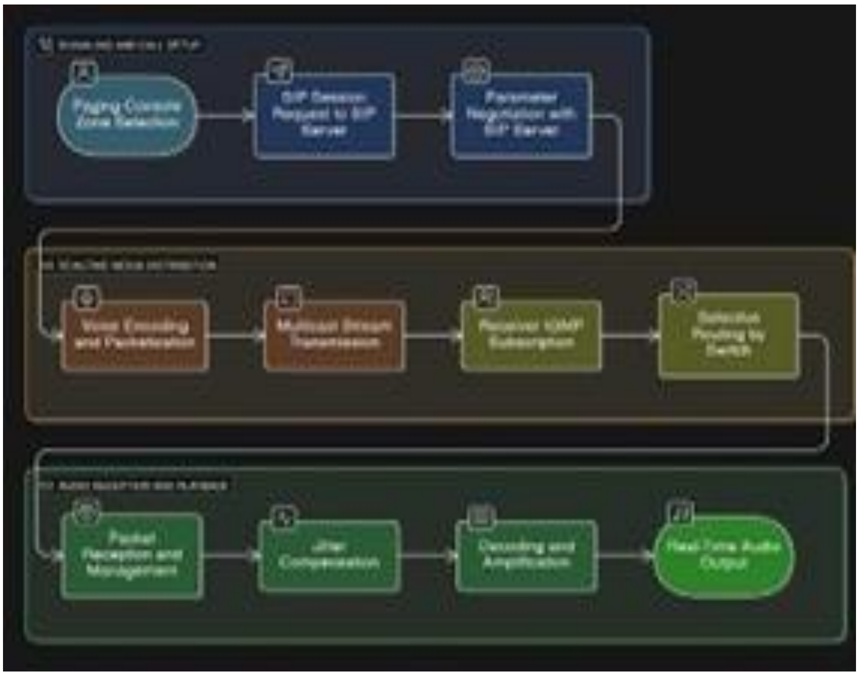
IV. OBJECTIVES

1. To design and implement a real-time IP-based broadcasting system for campus-level announcements and event coordination.
2. To achieve ultra-low latency communication ensuring instantaneous audio delivery across zones.
3. To integrate multicast RTP/IGMP protocols for efficient one-to-many communication without overloading the network.
4. To provide zone selectivity, allowing targeted or full- campus broadcasts based on event requirements.
5. Instant communication between administration and departments
6. Selective broadcasting (choose one, many, or all departments)
7. Clear, high-quality audio transmission
8. Reliable performance with low latency
9. Ease of use and management through a central control interface

With the help of this system, colleges can not only improve event coordination but also strengthen campus-wide communication infrastructure that can be reused for daily operations and safety alerts

V. IMPLEMENTATION DETAILS

A. System Architecture Diagram



a. Paging Console (Sender Unit): A desktop or embedded interface that captures voice input via a microphone and allows the user to select broadcasting zones (departments).

b. SIP Server: Acts as a control unit to manage session initiation, parameter negotiation (codec, sampling rate), and routing between sender and receivers.

c. Receiver Nodes: IP-based speakers or client devices connected to the campus LAN that receive and decode multicast audio streams for playback.

B. Software and Protocol Stack

a. Signaling Protocol: Session Initiation Protocol (SIP) handles connection establishment between sender and receivers.

b. Transport Layer: Real-Time Transport Protocol (RTP) over User Datagram Protocol (UDP) provides packet- based real-time transmission.

c. Group Management: Internet Group Management Protocol (IGMP) enables dynamic joining/leaving of multicast groups.

d. Codec Used: G.722 (wideband codec, 16-bit PCM, 16 kHz sampling) was selected for its low complexity and clear voice reproduction.

C. Data Flow and Operation

a. Zone: - The user selects one or multiple target departments through the paging console. Each department corresponds to a unique multicast IP address.

b. Session Initiation: - A SIP INVITE message is sent to the SIP server, which authenticates and negotiates parameters like codec and port numbers with receivers.

c. Audio Capture and Encoding: The live audio input from the microphone is digitized and encoded using the G.722 codec.

d. Packetization: - Encoded frames are divided into RTP packets with timestamps and sequence numbers for synchronization.

e. Multicast Transmission: - The RTP packets are transmitted to the network using IP multicast. Switches forward these packets only to the subscribed receivers.

f. Reception: - Receivers subscribed to the relevant multicast group use IGMP to join and receive the stream. They reorder packets, compensate for jitter, decode the audio, and play it through speakers.

D. Network Configuration

a. Network Type: Gigabit Ethernet LAN

b. Multicast IP Range: 239.1.1.10 – 239.1.1.99 (Department & All-Call Groups)

c. QoS Configuration: Audio packets prioritized via managed switches

d. Switch Settings: IGMP Snooping and Multicast VLAN Registration (MVR) enabled for efficient packet forwarding.

e. Latency Achieved: Average end-to-end latency measured below 30ms, ensuring near-instantaneous audio delivery.

VI. ADVANTAGES

- The system enables instantaneous transmission of voice messages across multiple departments or zones. The use of real-time protocols such as RTP (Real-Time Transport Protocol) ensures that announcements are delivered with minimal latency, which is crucial during live events or emergencies.
- By using advanced audio codecs such as G.722 or Opus, the system maintains high definition voice clarity even over standard network connections. This eliminates the distortion, noise, and echo issues commonly found in analog PA systems.
- The IP-based design allows easy addition of new departments or receivers without the need for extensive cabling or rewiring. Each new zone can simply subscribe to the appropriate multicast group
- IP multicast transmission, a single audio stream can be delivered to multiple receivers simultaneously. This drastically reduces network congestion compared to unicast systems, where separate streams would be needed for each receiver.

VI. APPLICATIONS

- Used for festival announcements, event coordination, and daily communication across departments.
- Enables emergency broadcasting during fire drills, evacuations, or important notices.
- Supports morning announcements, lecture updates, and examination alerts through centralized control.
- Can be deployed in corporate campuses for centralized announcements and meeting notifications.
- In industrial environments, it facilitates real-time alerts, safety warnings, and shift change notifications across production areas.
- Enhances communication efficiency in large factories or warehouses where manual announcements are impractical.
- Enables instant voice alerts for doctors, nurses, or emergency response teams.
- Can be integrated with hospital management systems to automatically announce emergencies such as ―Code Blue‖ or equipment failures.
- Provides silent or selective paging for specific departments like ICU, OPD, or Emergency.

VII. CONCLUSION

- The project ―Real-Time message Broadcasting for educational system successfully demonstrates the use of modern digital communication technologies to enhance campus-level coordination and event management. Traditional analog public address systems are gradually becoming outdated due to their limitations in coverage, flexibility, and scalability. By adopting an IP-based broadcasting approach, this project provides a cost- effective, efficient, and future-ready solution for real- time communication across multiple departments.
- The proposed system is designed to deliver instant voice announcements to specific zones or all departments simultaneously with high clarity and minimal latency. It integrates essential networking technologies such as Session Initiation Protocol (SIP), Real-Time Transport Protocol (RTP), and IP Multicast to ensure seamless and synchronized audio transmission. The use of selective zone paging allows the operator to communicate directly with intended groups without disturbing others, while multicast streaming ensures optimal bandwidth usage even in large networks.

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