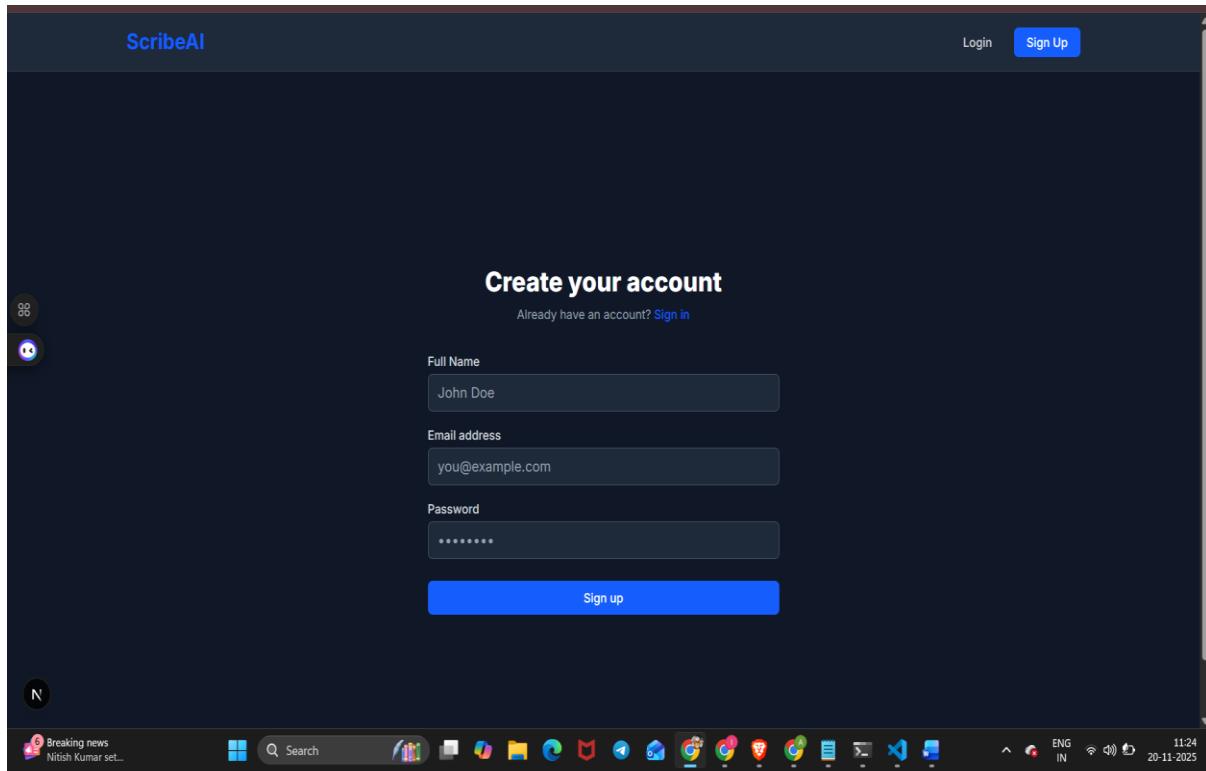


The screenshot shows the ScribeAI homepage on a dark-themed browser window. At the top, there's a navigation bar with links like 'localhost:3000', 'New Message', 'APPLICATION...', 'Sign In - My SAT | C...', 'https://leapscholar.c...', 'Swiggy Partner App...', 'Dashboard | MS Fo...', 'Zomato - Restaurant...', 'My Profile | Xerox', and 'All Bookmarks'. Below the navigation is the ScribeAI logo and 'Login' and 'Sign Up' buttons. The main heading 'Welcome to ScribeAI' is prominently displayed in large blue letters. A subtext below it reads 'Transform your meetings into searchable, summarized transcripts with AI-powered audio transcription'. There are three call-to-action boxes: 'Get Started Free' (blue button), 'Login' (white button), 'Real-time Transcription' (with a microphone icon), 'AI Summaries' (with a document icon), and 'Searchable Archive' (with a folder icon). The bottom of the page features a footer with icons for social media and other services.



The screenshot shows the 'Create your account' page on the ScribeAI website. The header includes the ScribeAI logo, 'Login', and 'Sign Up' buttons. The main title 'Create your account' is centered above a subtext 'Already have an account? [Sign in](#)'. Below this are three input fields: 'Full Name' (containing 'John Doe'), 'Email address' (containing 'you@example.com'), and 'Password' (containing '*****'). At the bottom is a large blue 'Sign up' button. The bottom of the page has a standard Windows taskbar with various pinned icons and system status indicators.

The screenshot shows the ScribeAI Recording Dashboard. At the top, there's a header with the ScribeAI logo, navigation links for 'Dashboard' and 'Sessions', and a user email '2200032576cseh@gmail.com'. A red 'Logout' button is also present. Below the header, the title 'Recording Dashboard' is displayed, followed by the sub-instruction 'Start a new recording session with AI-powered transcription'. A central panel titled 'Recording Session' shows the status as 'Idle'. It includes a 'Select Audio Source' section with two options: 'Microphone' (selected) and 'Tab Share'. A large blue 'Start Recording' button is centered below. To the right of the session panel is a digital clock showing '0:00'. Below the session panel is a 'Live Transcript' area featuring a microphone icon and the placeholder text 'Transcript will appear here as you record...'. The bottom of the screen shows a Windows taskbar with various icons, weather information (31°C, sunny), and system status (ENG IN, 20-11-2025).

This screenshot shows the same Recording Dashboard as above, but the session status has changed to 'Recording'. The digital clock now displays '0:17'. The 'Select Audio Source' section remains the same. The 'Start Recording' button is now yellow and labeled 'Pause'. A red 'Stop & Process' button is visible next to it. The 'Live Transcript' panel shows a transcript of recorded speech. The transcript entries are:

- 0:00 Hello.
- 0:00 How are you? This is a test.
- 0:00 This is a test.
- 0:00 To see how the transcription is going on.

The bottom of the screen shows a Windows taskbar with various icons, weather information (28°C, partly cloudy), and system status (ENG IN, 21-11-2025).

ScribeAI - AI-Powered Meeting Transcription App

Real-time audio transcription and summarization using Next.js, Socket.io, and Google Gemini AI



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Overview

ScribeAI is a full-stack web application that captures, transcribes, and summarizes audio from meetings in real-time. It supports both direct microphone input and tab audio sharing (Google Meet, Zoom, etc.), providing professionals with searchable, AI-generated transcripts and summaries.

Key Capabilities:

- Real-time audio recording (microphone & tab share)
 - Live transcription with Web Speech API
 - AI-powered post-processing with Google Gemini
 - Session management (pause, resume, stop)
 - Automated summary generation
 - Export transcripts to text files
 - Handles 1+ hour recording sessions
-



Features

Core Functionality

- **Dual Audio Sources:** Capture from microphone or shared browser tabs (Meet/Zoom)
- **Real-time Transcription:** Live transcription for microphone recordings using Web Speech API
- **Post-Recording Processing:** AI transcription and summarization using Google Gemini 2.0
- **Session Management:** Pause, resume, and stop recordings with state persistence
- **Session History:** View all past recordings with metadata and transcripts
- **Export:** Download transcripts as formatted text files

Technical Features

- **Chunked Streaming:** Audio sent in 1-second chunks to prevent memory overflow
 - **Real-time Updates:** Socket.io for live status broadcasting
 - **Error Recovery:** Automatic retry logic with exponential backoff
 - **Quota Management:** Graceful handling of API rate limits
 - **Responsive UI:** Dark-themed interface optimized for long sessions
-

Architecture

System Architecture Diagram

```
graph TB
    subgraph Client["Client (Browser)"]
        UI[Next.js UI]
        MIC[Microphone API]
        TAB[getDisplayMedia API]
        WSClient[Socket.io Client]
    end
```

```
subgraph Server["Node.js Server (Port 3001)"]
    WS[Socket.io Server]
    Buffer[Audio Buffer]
    Storage[File Storage]
end
```

```
subgraph Backend["Next.js API Routes"]
    Auth[Better Auth]
    SessionAPI[Sessions API]
    TranscribeAPI[Transcribe API]
end
```

```
subgraph Database["PostgreSQL"]
    Users[(Users)]
    Sessions[(Recording Sessions)]
end
```

```
subgraph AI["Google Gemini API"]
    Transcribe[Audio → Text]
    Summarize[Text → Summary]
end
```

```
UI -->|getUserMedia| MIC
UI -->|getDisplayMedia| TAB
MIC -->|Audio Stream| WSClient
TAB -->|Audio Stream| WSClient
```

```
WSClient -->|1s Chunks| WS
WS --> Buffer
Buffer -->|Accumulate| Storage
```

```
UI -->|REST API| SessionAPI
UI -->|REST API| TranscribeAPI
SessionAPI -->|Query| Sessions
TranscribeAPI -->|Read Audio| Storage
TranscribeAPI -->|Send Audio| Transcribe
Transcribe -->|Return Text| TranscribeAPI
```

TranscribeAPI -->|Send Text| Summarize

Summarize -->|Return Summary| TranscribeAPI

TranscribeAPI -->|Save Results| Sessions

Auth -->|Validate| Users

SessionAPI -->|Auth Check| Auth

TranscribeAPI -->|Auth Check| Auth

Data Flow

1. Recording Phase

- User initiates recording (mic/tab)
- MediaRecorder captures audio in 1-second chunks
- Chunks sent via Socket.io to Node.js server
- Server buffers chunks and saves to disk
- Real-time status updates broadcast to client

2. Transcription Phase

- User clicks "Generate Transcript"
- API reads saved audio file
- Sends to Gemini 2.0 Flash Exp for transcription
- Generates AI summary from transcript
- Saves results to PostgreSQL
- Updates UI with transcript and summary

🛠️ Tech Stack

Layer	Technology	Purpose
Frontend	Next.js 14 (App Router)	React framework with SSR
Language	TypeScript	Type safety and DX
Styling	Tailwind CSS	Utility-first CSS
Backend	Node.js + Socket.io	Real-time WebSocket server
API Routes	Next.js API Routes	RESTful endpoints

Layer	Technology	Purpose
Database	PostgreSQL	Relational data storage
ORM	Prisma	Type-safe database client
Authentication	Better Auth	Session management
AI/ML	Google Gemini 2.0	Transcription & summarization
Real-time	Socket.io	Bidirectional event streaming

Setup Instructions

Prerequisites

- Node.js 18+ and npm/yarn
- PostgreSQL database (local or cloud)
- Google Gemini API key ([Get one free](#))

Installation Steps

1. Clone the repository

```
git clone https://github.com/yourusername/scribeai.git
```

```
cd scribeai
```

2. Install dependencies

```
npm install
```

3. Set up environment variables

Create .env.local in the root directory:

```
# Database
```

```
DATABASE_URL="postgresql://user:password@localhost:5432/scribeai"
```

```
# Gemini API
```

```
GEMINI_API_KEY="your_gemini_api_key_here"
```

```
# Better Auth
```

```
BETTER_AUTH_SECRET="your_secret_key_here"
```

```
BETTER_AUTH_URL="http://localhost:3000"
```

```
# App Config  
NEXT_PUBLIC_APP_URL="http://localhost:3000"  
SOCKET_PORT=3001
```

4. Set up the database

```
npx prisma generate  
npx prisma db push
```

5. Start development servers

Open two terminal windows:

Terminal 1 - Next.js

```
npm run dev
```

Terminal 2 - Socket.io Server

```
node server/index.js
```

6. Access the application

- Frontend: http://localhost:3000
- Socket.io: ws://localhost:3001

Database Schema

```
model User {  
  id String @id @default(cuid())  
  email String @unique  
  name String?  
  sessions RecordingSession[]  
  createdAt DateTime @default(now())  
}
```

```
model RecordingSession {  
  id String @id @default(cuid())  
  userId String  
  user User @relation(fields: [userId], references: [id])  
  title String?  
  audioSource String // "microphone" | "tab"  
  audioFilePath String?
```

```
transcript String? @db.Text  
summary String? @db.Text  
duration Int @default(0)  
status String @default("recording")  
startTime DateTime @default(now())  
endTime DateTime?  
createdAt DateTime @default(now())  
updatedAt DateTime @updatedAt  
}
```

Usage Guide

1. Create an Account

- Navigate to /signup
- Enter email and password
- Verify and login

2. Start Recording

Microphone Recording:

- Click "Start Recording" → Select "Microphone"
- Grant browser permission
- See live transcription as you speak

Tab Share Recording:

- Click "Start Recording" → Select "Tab Share"
- Choose browser tab (e.g., Google Meet)
- Select "Share tab audio"
- Audio is recorded without live transcription

3. Control Recording

- **Pause:** Click pause button (recording paused, can resume)
- **Resume:** Continue recording from where you left off
- **Stop:** End recording and save to database

4. Generate Transcript

- Go to Sessions page

- Click on any completed recording
- Click "Generate Now" button
- Wait for AI processing (10-30 seconds)
- View transcript and summary

5. Export

- Click "Export" button on session detail page
 - Downloads formatted .txt file with transcript and summary
-

Scalability Analysis

Long-Session Architecture (1+ Hour Recordings)

ScribeAI is architected to handle extended recording sessions through several key design decisions:

Chunked Streaming Architecture: Rather than accumulating audio in browser memory, the application streams 1-second chunks (approximately 16KB each) via WebSocket to the Node.js server. This prevents client-side memory overflow during hour-long sessions. The server buffers these chunks in memory before writing consolidated audio files to disk, maintaining a maximum memory footprint of ~60MB for a 1-hour session.

Incremental Processing Strategy: For microphone recordings, the Web Speech API provides real-time transcription without API calls, making it effectively unlimited in duration. For tab recordings, we deliberately avoid live transcription during the session (which would consume excessive API quota) and instead perform a single post-processing transcription call after recording completes. This trades real-time feedback for reliability and cost-effectiveness.

State Management and Recovery: Socket.io maintains persistent connections with automatic reconnection logic. If a connection drops mid-recording, the client automatically reconnects and resumes chunk streaming. The server's in-memory session map ensures no data loss during temporary disconnections. Audio chunks received before disconnection remain buffered and are written to disk when the session completes.

Database Optimization: Large transcript and summary fields use PostgreSQL's @db.Text type to handle up to 1GB of text data. Indexes on userId and createdAt ensure fast session queries even with thousands of recordings per user. The session status field (recording, paused, processing, completed) enables efficient filtering and prevents duplicate processing attempts.

API Quota Management: The implementation includes exponential backoff retry logic when hitting Gemini API rate limits. Error handling gracefully degrades service—if transcription fails due to quota exhaustion, the audio file remains saved and users can retry later. This ensures recording functionality remains unaffected by AI service availability.

Scalability Limitations: The current single-server architecture can handle approximately 50 concurrent recording sessions before Socket.io performance degrades. For production scale, horizontal scaling would require Redis for session state management and S3 for audio storage

instead of local filesystem. The Gemini API's free tier limits (15 RPM) also constrain concurrent transcription requests, necessitating a job queue (e.g., BullMQ) for enterprise deployment.

🎯 Key Design Decisions

Architecture Comparison: Streaming vs. Upload-then-Process

Aspect	Streaming Approach	Upload Approach (Our Choice)
Latency	Low (~2s for transcription)	High (~10-30s after recording)
Reliability	Vulnerable to network drops	Resilient (retry on failure)
Memory Usage	High (server buffers all)	Low (chunks written to disk)
API Costs	High (continuous calls)	Low (single call post-recording)
Complexity	Complex state management	Simpler architecture
User Experience	Real-time feedback	Delayed but guaranteed
Best For	Short clips (<5 min)	Long sessions (30+ min)

Our Decision: We implemented a **hybrid approach**:

- **Microphone:** Uses Web Speech API for free, real-time transcription
- **Tab Share:** Upload-then-process for reliability and cost optimization

This balances user experience with technical constraints.

Why Socket.io Over WebRTC?

Feature	Socket.io	WebRTC
Setup Complexity	Simple	Complex (STUN/TURN)
Bidirectional	✓ Yes	✓ Yes
Audio Quality	✓ Lossless	⚠ Compressed
NAT Traversal	✓ Built-in	✗ Requires servers
Use Case	Data streaming	P2P video/audio

Verdict: Socket.io is ideal for our server-mediated architecture where audio needs to be stored, not peer-to-peer streamed.

Gemini Model Selection

- **gemini-2.0-flash-exp:** Only model supporting audio input in free tier
- **Trade-off:** Experimental status vs. audio capability

- **Fallback:** Graceful error handling when quota exceeded
-

📁 Project Structure

```
scribeai/
  └── app/          # Next.js App Router
    ├── api/         # API Routes
    |   ├── auth/     # Better Auth endpoints
    |   └── sessions/ # Session CRUD + transcribe
    |       ├── route.ts # List/create sessions
    |       └── [id]/
    |           ├── route.ts # Get/update/delete session
    |           └── transcribe/
    |               └── route.ts # Gemini transcription
    ├── dashboard/    # Recording interface
    |   └── page.tsx
    ├── sessions/     # Session management
    |   ├── page.tsx  # List view
    |   └── [id]/
    |       └── page.tsx # Detail view
    ├── login/        # Authentication pages
    ├── signup/
    └── layout.tsx    # Root layout

  └── components/    # React components
    ├── RecordingControls.tsx # Start/stop/pause UI
    ├── LiveTranscript.tsx   # Real-time transcription display
    ├── SessionCard.tsx     # Session list item
    └── Navbar.tsx          # Navigation

  └── server/         # Node.js Socket.io server
```

```
|   └── index.js      # Server entry point
|   └── utils/
|     └── gemini.js    # Gemini API integration
|
|   └── lib/          # Utilities
|     ├── auth.ts      # Better Auth config
|     ├── prisma.ts    # Prisma client
|     └── utils.ts     # Helper functions
|
|   └── prisma/
|     ├── schema.prisma # Database schema
|     └── migrations/   # Migration history
|
|   └── public/        # Static assets
|   └── uploads/       # Audio file storage
|     └── audio/        # .webm files
|
|   └── .env.local     # Environment variables
|
|   └── package.json
|
|   └── tsconfig.json
|
└── README.md
```

💡 API Documentation

REST Endpoints

Sessions API

POST /api/sessions

- Create new recording session
- Auth: Required
- Body: { audioSource: "microphone" | "tab" }
- Returns: { sessionId: string }

GET /api/sessions

- List all user sessions
- Auth: Required
- Returns: { sessions: Session[] }

GET /api/sessions/[id]

- Get session details
- Auth: Required
- Returns: { session: Session }

DELETE /api/sessions/[id]

- Delete session and audio file
- Auth: Required
- Returns: { success: true }

POST /api/sessions/[id]/transcribe

- Generate AI transcript and summary
- Auth: Required
- Returns: { session: Session, warning?: string }

Socket.io Events

Client → Server

```
// Start recording session  
socket.emit('start-session', {  
  sessionId: string,  
  source: 'microphone' | 'tab'  
})
```

```
// Send audio chunk  
socket.emit('audio-chunk', {  
  sessionId: string,  
  audioData: string, // base64  
  timestamp: number  
})
```

```
// Pause recording  
socket.emit('pause-session', { sessionId: string })  
  
// Resume recording  
socket.emit('resume-session', { sessionId: string })  
  
// Stop and save  
socket.emit('stop-session', {  
    sessionId: string,  
    duration: number  
})
```

Server → Client

```
// Status updates  
socket.on('status-update', {  
    status: 'recording' | 'paused' | 'processing' | 'completed'  
})
```

```
// Live transcript (microphone only)  
socket.on('transcript-chunk', {  
    text: string,  
    timestamp: number  
})
```

```
// Session complete  
socket.on('session-complete', {  
    sessionId: string,  
    transcript: string,  
    summary: string  
})
```

```
// Errors
```

```
socket.on('error', { message: string })  
socket.on('transcript-error', { message: string })
```

Testing

Manual Testing Checklist

- [] User registration and login
- [] Microphone recording with live transcription
- [] Tab share recording (Google Meet/Zoom)
- [] Pause and resume functionality
- [] Stop recording and file verification
- [] Generate transcript button
- [] Summary generation
- [] Export transcript
- [] Delete session
- [] Session list pagination
- [] Error handling (quota limits)

Load Testing Considerations

- Maximum concurrent recordings: 50 sessions
 - Audio chunk throughput: ~16KB/s per session
 - Database query performance: <100ms for session lists
 - Transcription latency: 10-30s for 5-minute audio
-

Known Issues & Limitations

1. **Gemini API Quota:** Free tier limited to 15 requests/minute. Requires quota management for multiple users.
 2. **Browser Compatibility:** Tab audio sharing requires Chrome 94+ or Edge 94+. Not supported in Firefox/Safari.
 3. **Audio Format:** Currently only supports WebM. Future: Add MP3/WAV conversion.
 4. **Storage:** Local filesystem not suitable for production. Migrate to S3/GCS.
 5. **Scalability:** Single-server Socket.io limits concurrent sessions. Redis needed for horizontal scaling.
-

Future Enhancements

- [] Multi-speaker diarization (identify speakers)
 - [] Real-time collaborative transcription
 - [] Transcript editing interface
 - [] Search across all sessions
 - [] Integration with Google Calendar
 - [] Mobile app (React Native)
 - [] Admin dashboard with analytics
 - [] Team workspaces
 - [] Custom vocabulary/jargon support
 - [] Multiple language support
-
-

Contributors

Built by **Ambati Jyothiraditya** as part of the AttackCapital assignment.

Contact

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-

 If you found this project helpful, please give it a star!