# Ticket 2: ABC-2851

**Status:** In Progress
**Priority:** High
**Summary:** Severe voice quality degradation with 400ms+ latency and packet loss in multi-branch deployment
**Reporter:** Michael Rodriguez (TechFlow Solutions)
**Assignee:** Jennifer Park (Voice Engineering)
**Created Date:** June 3, 2025
**Updated Date:** June 7, 2025

## Description

**Customer-Reported Symptoms:** TechFlow Solutions reports significant voice quality issues across their 8-branch deployment. Callers experience robotic/distorted audio, frequent audio dropouts lasting 2-3 seconds, and delayed responses from the Voice Receptionist. The issue is most pronounced in their West Coast branches (San Francisco, Los Angeles, Seattle) but also affecting Chicago and Miami locations.

**Error Messages/Logs:**

```
[2025-06-03 14:15:22] WARN AudioProcessor: High jitter detected - 1
[2025-06-03 14:15:23] ERROR RTPHandler: Packet sequence gap detecte
[2025-06-03 14:15:24] WARN VoiceQualityMonitor: MOS score below thr
[2025-06-03 14:15:25] ERROR AudioCodec: Buffer underrun - Insuffici
[2025-06-03 14:15:26] INFO NetworkQoS: Adaptive bitrate triggered -
```

**Environmental Details:** - Browser: Edge 124.0.2478.67 (corporate standard) - Network Infrastructure: Multi-site MPLS network with QoS enabled - Bandwidth per site: 100 Mbps shared across all applications - WebRTC configuration: STUN/TURN servers in us-east-1 - Audio codec: G.722 (primary), G.711 (fallback) - Geographic distribution: 8 sites across 6 time zones - Average RTT to voice servers: 89ms (West Coast), 34ms (East Coast)

**Impact on Business Operations:** - 34% increase in call transfer requests to human operators - Customer complaints about "unprofessional automated system" - Regional sales team reporting lost prospects due to poor first impression - IT helpdesk receiving 15-20 voice quality tickets per day - Considering rollback to previous phone system

**Current Investigation Status:** - Network analysis completed: Identified asymmetric routing causing 180ms additional latency on West Coast paths - Packet capture analysis shows 3.2% packet loss during business hours vs. 0.1% after hours - QoS configuration review revealed voice traffic not properly prioritized over MPLS links - WebRTC TURN server logs indicate suboptimal candidate selection

**Attempted Solutions:** 1. **Codec optimization:** Switched primary codec from G.722 to G.711u for affected sites - marginal improvement (MOS increased from 2.1 to 2.4) 2. **Buffer adjustments:** Increased jitter buffer size from 100ms to 200ms - reduced dropouts by 15% but increased latency 3. **TURN server relocation:** Deployed additional TURN servers in us-west-1 - improved West Coast RTT by 23ms 4. **Network path optimization:** Working with ISP to implement symmetric routing - pending completion

**Next Steps:** - Deploy dedicated voice VLAN configuration across all sites (scheduled June 9) - Implement adaptive jitter buffer with ML-based optimization (June 11) - Coordinate with network team for QoS policy updates (June 12) - Conduct end-to-end voice quality testing with updated configuration (June 13) - Customer validation testing scheduled for June 14-15