JETIR.ORG ISSN: 2349-5162 | ESTD Year : 2014 | Monthly Issue
JOURNAL OF EMERGING TECHNOLOGIES AND INNOVATIVE RESEARCH (JETIR)
An International Scholarly Open Access, Peer-reviewed, Refereed Journal

Voice Quality Technology and Testing: Advancements, Techniques, and Applications

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Abstract : Voice Quality Testing refers to the process of evaluating and ensuring the clarity, intelligibility, and overall quality of audio (voice) in telecommunication systems, including phone calls, interactive voice response (IVR) systems, and other voice based applications. The goal of voice quality testing is to ensure that the system is delivering clear, understandable and pleasant audio experiences for users. IVR systems are commonly used in customer service and support, so any issues with voice quality can impact user satisfaction and operational efficiency.

***IndexTerms* – Voice Quality,Software Testing , PESQ, POLQA, MOS scoring**

I. Introduction

Voice quality plays a critical role in the effectiveness of a Customer Service Interactive Voice Response (IVR) system for several reasons. The voice quality of an IVR system is often the first point of contact for customers. A clear, warm, and professional voice creates a positive first impression, making customers feel comfortable and confident that they will receive the assistance they need. A high-quality voice ensures that customers can easily understand the instructions and options presented by the IVR system. If the voice is clear, articulate, and pleasant, customers are less likely to experience frustration, which leads to a more positive interaction. A poor or muffled voice can cause confusion, miscommunication, or the need for customers to repeat themselves, which negatively impacts the customer experience. The tone of the voice influences how customers feel about their experience. A calm, friendly, and empathetic voice can help ease frustration, especially when customers are dealing with complex issues or are in distress. voice quality is essential in creating an efficient, professional, and enjoyable customer service experience. Clear and pleasant voice prompts not only help guide customers but also foster trust and satisfaction, which can positively influence customer loyalty and retention

II. Key Components of Voice Quality Testing in IVR

1. **Clarity and Intelligibility:**

a. **Clear Speech**: The voice prompts should be crisp, without distortion, so that callers can easily understand the information being provided.
b. **Pronunciation**: Ensure proper pronunciation of words, including local accents or specific terminology. This is particularly important for multilingual IVR systems.

2. **Volume Level and Consistency:**

a. **Balanced Volume**: The speech should be loud enough to be heard clearly but not too loud to cause discomfort.
b. **Consistent Volume**: The volume should remain consistent across different prompts and throughout the IVR flow. Sudden changes in volume can be jarring for the user.

3. **Naturalness and Tone:**

a. **Natural Sounding Voice**: The IVR voice should sound conversational, avoiding robotic or monotone delivery. A natural-sounding voice improves user engagement.
b. **Tone and Emotion**: The tone of the voice should be appropriate for the scenario (e.g., friendly and empathetic in customer service, neutral and professional for transactional queries)

4. **Background Noise:**

a. **Clean Audio**: There should be minimal to no background noise or interference. Any noise, such as hums, clicks, or static, can reduce the clarity of the speech.
b. **Echo/Feedback**: Make sure there is no echo or feedback issues that could interfere with the caller's experience.

5. **Speech Recognition and Accuracy:**

a. **Recognition Testing**: If the IVR system uses speech recognition to capture user inputs, the clarity of prompts should be tested in relation to the system's ability to accurately understand user speech.
b. **Noise Tolerance**: Test how well the speech recognition works in noisy environments, where background sounds (like traffic or office noise) might interfere with recognition.

6. **Latency and Response Time:**

a. **Prompt Timing**: Measure the time between when the caller speaks or presses a key and when the IVR responds. Delays can cause confusion or frustration.

b. **Seamless Transitions**: Test how quickly the system transitions between different prompts or when switching to an agent.

7. **Error Handling and Recovery:**

a. **Handling Misunderstandings**: Ensure that if a user's input is misunderstood (either through speech recognition or DTMF), the system can handle the error gracefully, offering a chance for the user to try again or offer alternative options.

b. **Clear Error Messages**: If an error occurs (such as an invalid input or system failure), the system should provide clear instructions on what the user should do next.

8. **Multiple Languages (if applicable):**

a. **Multilingual Support**: If the IVR system supports multiple languages, testing should ensure that the voice quality is consistent and clear across all language options.

b. **Cultural Sensitivity**: The tone and phrasing should be culturally appropriate for the specific regions or audiences the system serves.

9. **Volume Normalization (if applicable):**

a. Ensure that if the system includes multiple voices (e.g., for different sections or steps in the IVR process), the volume and tone of the voices are normalized so that the transitions feel smooth and natural.

III. Testing Methods

1. **Automated Testing Tools:**

a. Use automated voice testing tools that simulate various environments (e.g., different network conditions, mobile phones, or low-quality connections) to test the clarity, latency, and intelligibility of the IVR system.

b. Voice Quality Analysis Tools: These tools measure parameters like signal-to-noise ratio (SNR), Mean Opinion Score (MOS), jitter, and packet loss for voice quality analysis.

2. **Real User Testing (User Experience Testing):**

a. Conduct user testing with real customers or focus groups to assess the naturalness, clarity, and intelligibility of the IVR prompts in real-world conditions.[2]

b. Gather feedback on how easy it is for users to interact with the IVR system, especially when the system uses speech recognition or offers multiple options.

3. **Subjective Testing (MOS Scores):**

a. **Mean Opinion Score (MOS)**: MOS is a widely used measure for subjective testing of voice quality, where testers rate the quality of speech on a scale from 1 to 5. A higher MOS score indicates better quality.

4. **Network Conditions Testing:**

a. Test the IVR system under different network conditions, such as poor or fluctuating signal strength, to evaluate how voice quality holds up during real-time calls.

5. **End-to-End Call Flow Testing:**

a. Perform comprehensive testing of the entire IVR call flow, from the initial greeting to the final step, ensuring that all parts of the system work seamlessly and that the voice quality remains consistent.[7]

Common Challenges in IVR Voice Quality:

- **Network Issues**: Poor network conditions can affect voice clarity, leading to dropped calls or distorted speech.
- **Compression**: Audio compression (often used to save bandwidth) can sometimes degrade the quality of the voice prompts.
- **Device Compatibility**: IVR systems should be tested across various devices (e.g., landlines, mobile phones, VoIP services) to ensure consistent voice quality.[8]

IV. Technologies for Voice Quality Testing

1. **PESQ (Perceptual Evaluation of Speech Quality):**

PESQ (Perceptual Evaluation of Speech Quality) is an objective testing method used to assess the quality of speech in telecommunication systems, including voice-over-IP (VoIP), mobile networks, and IVR systems [4]. PESQ provides a way to evaluate the perceptual quality of speech transmission, comparing a reference signal (unprocessed, original speech) with a processed signal (the speech that has passed through the communication system). This method is based on the human perception of speech quality, making it highly useful for predicting how a listener will perceive the audio quality in real-world conditions. PESQ was developed as a standard by the International Telecommunication Union (ITU) in 2001 and is widely used to evaluate voice quality in a variety of contexts, such as call centers, mobile communication, VoIP services, and even network equipment performance

How PESQ Works:

PESQ evaluates the subjective quality of speech by comparing the original (reference) audio with the processed (test) audio that has passed through a communication system. The process is modeled based on human auditory perception, so the score is indicative of how the average listener would rate the quality of the call.

Here's how PESQ works in more detail:

1. **Reference Signal (Original Audio):**
   a. This is the original unprocessed speech that would have been recorded or transmitted without any degradation. It's typically captured from the source before transmission (for example, a clean audio file of a speaker saying a sentence).
2. **Test Signal (Processed Audio):**
   a. The speech signal is then processed through a communication channel, such as a telecommunication network, a VoIP service, or an IVR system. This processed signal might undergo compression, network packet loss, jitter, distortion, or other quality-degrading factors.
3. **Perceptual Model:**
   a. PESQ uses a perceptual model that mimics how human listeners perceive differences in speech. This model accounts for various factors like:
      i. **Loudness**: Volume level and dynamic range of the speech.
      ii. **Clarity**: How intelligible the speech sounds (no distortion, noise, etc.).
      iii. **Spectral Distortion**: How much the frequency spectrum of the speech signal is altered.
      iv. **Delay**: Time lag between the original and processed signals.
      v. **Packet Loss**: If using VoIP, how missing packets affect the audio stream.
      vi. **Noise**: Background noise or artifacts added to the signal.
4. **PESQ Algorithm:**
   a. The algorithm compares the original and processed signals based on time alignment, frequency analysis, and distortion models. It computes a score that reflects how the listener would subjectively rate the speech quality, accounting for factors like signal degradation (e.g., compression artifacts, noise, jitter).
5. **Score Calculation:**
   a. The output of PESQ is a score that reflects the Mean Opinion Score (MOS) on a scale from 1 to 5.
      i. **5**: Excellent quality (e.g., as if the speaker and listener were in the same room).
      ii. **4**: Good quality (clear, minimal distortion or noise).
      iii. **3**: Fair quality (noticeable degradation, but intelligible).
      iv. **2**: Poor quality (hard to understand, severe degradation).
      v. **1**: Bad quality (unintelligible, speech may be garbled or impossible to understand).

PESQ's MOS Score Calculation:

While PESQ calculates a detailed perceptual quality score based on various factors (distortion, noise, delay), it ultimately outputs a MOS score, which is typically used in speech quality assessments.

- **MOS (Mean Opinion Score)**: This is a score based on human listeners' subjective opinions of the audio quality, averaged across several people. The PESQ score is calculated to closely match this subjective rating, making it easier to compare with other MOS ratings from human listeners.
- **PESQ MOS Range**:
  - **MOS 4.0–5.0**: Excellent, no noticeable quality issues.
  - **MOS 3.0–3.9**: Acceptable, with minor degradation.
  - **MOS 2.0–2.9**: Poor quality, likely noticeable issues.
  - **MOS < 2.0**: Very poor, nearly unusable quality

Advantages of PESQ:

1. **Objective and Reliable:**
   a. PESQ provides an objective measurement of voice quality, based on algorithms that simulate human perception. This is highly useful for testing across large-scale networks or automated testing scenarios, where subjective testing by human listeners is impractical.
2. **ITU Standardized:**
   a. As a standard recognized by the International Telecommunication Union (ITU), PESQ provides a uniform methodology for comparing speech quality across different systems, ensuring consistency in results.
3. **Versatile Across Networks:**
   a. PESQ can be applied across various telecommunications networks and voice technologies, including traditional telephone lines, mobile networks, and VoIP systems. This makes it versatile for evaluating performance in diverse environments.
4. **Predictive of Real-World Experience:**
   a. Since PESQ is designed to reflect human auditory perception, its scores are predictive of what real-world users will experience when interacting with an IVR, VoIP call, or mobile phone system.
5. **Supports Automated Testing:**
   a. PESQ allows for automated testing of speech quality, enabling continuous monitoring and benchmarking of systems without the need for human testers in every case.

Limitations of PESQ:

1. **Narrowband Focus:**
   a. PESQ was originally developed for narrowband speech (300 Hz to 3,400 Hz). While there have been extensions to support wideband audio (e.g., POLQA), PESQ may not fully capture the perceptual differences in higher-bandwidth systems that use full-band or super-wideband codecs (such as those found in modern mobile and VoIP systems).[3]
2. **Sensitivity to Distortion Type:**
   a. PESQ may be less sensitive to certain types of audio distortion. For instance, PESQ might not detect echo or delay as well as it detects noise or clipping, because these factors can be harder to quantify based on the human auditory model.
3. **Fixed Test Conditions:**
   a. PESQ assumes that both the reference and the test signal are synchronized and that the same speech material is being tested. Variability in the content or recording conditions could affect the score, meaning real-world variability (e.g., a noisy environment or an atypical accent) might not always align perfectly with the PESQ results.
4. **Limited to Speech:**
   a. PESQ is focused purely on speech quality and may not be suitable for testing music or other non-speech audio content. Therefore, it is not universally applicable to all types of media.
5. **Not Real-Time:**
   a. While PESQ is highly effective for post-call analysis, it may not be suitable for real-time speech quality assessment during an active call. For live testing, other tools like POLQA or network monitoring systems may be preferred.

PESQ is a powerful and widely adopted tool for objective voice quality testing, offering a reliable and automated method for evaluating speech clarity, intelligibility, and overall performance in telecommunications systems. It plays an essential role in testing systems like IVRs, VoIP services, and call centers, ensuring that end users receive clear and high-quality audio during their interactions. By providing an objective MOS score, PESQ can help service providers, network operators, and quality assurance teams quickly identify and address voice quality issues that could affect customer experience.

2. **POLQA (Perceptual Objective Listening Quality Assessment) :**

POLQA (Perceptual Objective Listening Quality Assessment) is a more advanced version of PESQ (Perceptual Evaluation of Speech Quality), designed to address some of the limitations of PESQ and provide higher accuracy for modern, wideband, and HD voice systems. It was introduced by the International Telecommunication Union (ITU) as the standard for evaluating speech quality in voice communication systems, particularly those with wideband and super-wideband codecs, such as VoLTE, VoIP, and HD Voice services.[1]

POLQA builds on the foundation laid by PESQ but offers several significant enhancements, making it more suitable for modern telecommunication networks, including mobile networks (e.g., 4G/5G), VoIP services, and next-generation audio codecs.POLQA is an objective speech quality measurement tool designed to predict the quality of speech from a user's perspective. It does this by comparing a reference signal (original, unprocessed speech) to a test signal (processed speech, typically passed through a telecommunications network or audio codec).POLQA evaluates the perceived speech quality based on human hearing models and provides a score that corresponds to a Mean Opinion Score (MOS), a rating scale used in telecommunications to gauge audio quality.

How POLQA Works:

POLQA operates in a similar manner to PESQ but has been specifically developed to handle the increased complexity and high-quality requirements of wideband (15 kHz), super-wideband (20 kHz), and even full-band (50 kHz) audio signals. It accounts for a range of factors that can degrade or distort the voice signal, such as packet loss, jitter, delay, and codec artifacts, and models how these would be perceived by a human listener.

Here's a breakdown of the steps in POLQA testing:

1. **Reference Signal (Original Audio):**
   a. The original audio is captured in high quality, often recorded in ideal conditions. This can be speech from a clean audio file or a conversation with no processing or degradation (i.e., the "raw" audio).
2. **Test Signal (Processed Audio):**
   a. The processed test signal is the same audio after it has passed through a telecommunications channel, such as a VoIP network, mobile network, or any other voice transmission system that could degrade the signal (compression, packet loss, jitter, echo, noise, etc.).
3. **Pre-Processing and Time Alignment:**
   a. POLQA time-aligns the reference and test signals to ensure that they are synchronized. Since network-induced delays and packet loss can cause temporal misalignment, this step ensures that the algorithm compares the correct portions of the signals.
4. **Perceptual Modeling:**
   a. POLQA uses perceptual models of human hearing to simulate how the test signal would be perceived by a listener. These models account for:
      i. **Loudness and Dynamics**: Human listeners are sensitive to loudness fluctuations, and POLQA evaluates how well the dynamics of the signal are preserved.

ii. **Spectral Distortion**: POLQA checks if the frequency range (spectrum) of the speech has been distorted due to bandwidth restrictions or codec artifacts.[10]
iii. **Noise**: Background noise or distortion in the processed signal is evaluated for its impact on intelligibility and clarity.
iv. **Artifacts**: POLQA assesses artifacts caused by network issues, such as jitter, packet loss, or compression artifacts.
v. **Echo and Delay**: POLQA takes into account how delay or echo might impact the perceived quality of speech.

5. **Score Calculation:**
   a. After comparing the reference and test signals, POLQA generates a score that corresponds to a Mean Opinion Score (MOS), typically ranging from 1 (bad) to 5 (excellent). The score reflects how users would rate the quality of the voice signal after being processed.

POLQA MOS Score Scale:

The MOS scale used in POLQA, similar to PESQ, indicates the overall perceived quality of the speech:

- **MOS 5.0**: **Excellent** quality (no perceptible issues, ideal listening experience).
- **MOS 4.0–4.9**: **Good** quality (minor distortions or noise but still clear and intelligible).
- **MOS 3.0–3.9**: **Fair** quality (noticeable degradation, some clarity loss, but still understandable).
- **MOS 2.0–2.9**: **Poor** quality (audio is hard to understand, significant distortions or noise).
- **MOS 1.0–1.9**: **Bad** quality (unintelligible, severe distortion, may be completely unrecognizable).[3]

The higher the MOS score, the better the quality of the speech signal, and the closer it is to a real-world listening experience.

Key Improvements of POLQA Over PESQ:

1. **Wideband and Super-Wideband Support:**
   a. POLQA was specifically designed to support wideband (15 kHz) and super-wideband (20 kHz) codecs, which are increasingly common in modern communication networks, especially in HD Voice systems. PESQ, on the other hand, was optimized for narrowband signals (300–3,400 Hz) and doesn't accurately reflect the performance of wideband or high-definition audio.
   b. POLQA's support for high-definition audio makes it a better fit for evaluating VoLTE (Voice over LTE), VoWiFi (Voice over Wi-Fi), and other modern communication systems that deliver HD voice.
2. **Support for Higher Quality Audio:**
   a. POLQA can evaluate full-band speech (up to 50 kHz), meaning it can accurately assess the quality of calls made using high-fidelity codecs such as AMR-WB or Opus. This is essential for systems like VoIP and next-generation mobile networks that aim to provide high-quality audio even in challenging conditions.
3. **Better Handling of Delays, Echo, and Jitter:**
   a. POLQA includes improved algorithms to handle the effects of network delay, echo, jitter, and packet loss in the test signal. This is critical because modern communication networks often experience variable network conditions that impact call quality, and POLQA is able to simulate the way these factors influence human listeners' perceptions of the voice quality.
4. **Improved Accuracy for Post-Processing Effects:**
   a. POLQA is better at detecting post-processing artifacts such as noise suppression, echo cancellation, and bandwidth limitations that could degrade voice quality in modern telecommunication systems.
5. **ITU-T Recommendation (P.863):**
   a. POLQA is standardized under the ITU-T P.863 recommendation, which ensures that it meets international standards for objective voice quality testing. This makes it a widely accepted and trusted method for evaluating voice quality across a range of applications and industries.[9]

Advantages of POLQA:

1. **Accurate Representation of Human Perception:**
   a. By using advanced perceptual modeling, POLQA provides a highly accurate prediction of how real users would experience voice quality. It accounts for the limitations and distortions that humans are most sensitive to, such as speech clarity, background noise, and distortion.
2. **Wide Range of Applications:**
   a. POLQA is highly effective for testing VoIP systems, mobile networks (including VoLTE/VoWiFi), call centers, and IVR systems. It is used by service providers, network operators, and equipment manufacturers to evaluate and benchmark the performance of communication systems.
   b. Network Operators use POLQA to monitor network performance, optimize voice services, and ensure that high-quality voice calls are consistently delivered to end users.
3. **High-Fidelity Support:**
   a. POLQA's ability to evaluate HD voice, wideband, and super-wideband audio ensures that it is a future-proof tool for assessing the increasing complexity of modern voice codecs and communication systems.
4. **Scalable and Automated:**

a. POLQA can be implemented in automated testing systems, making it highly scalable for large-scale network performance monitoring or quality assurance testing across multiple devices or platforms. This is particularly useful in environments where real-time or continuous monitoring of voice quality is required.

5. **Realistic Testing in Real-World Conditions:**
   a. POLQA can be used to evaluate systems under real-world network conditions, including packet loss, jitter, and varying latency, allowing it to deliver insights into how systems perform under typical network stresses.

Limitations of POLQA:

1. **Requires High-Quality Reference Signals:**
   a. For accurate results, POLQA requires high-quality reference signals to compare against the processed signals. The accuracy of the MOS score depends on the quality of the original signal[5].
2. **Complexity:**
   a. POLQA is more complex and computationally intensive than PESQ due to its support for high-definition audio, and this can require more processing power, particularly for large-scale testing.
3. **Not Always Suitable for Non-Speech Content:**
   a. Like PESQ, POLQA is designed specifically for speech quality and may not be suitable for testing non-speech audio content (e.g., music or multimedia files).
4. **Post-Call Analysis**:
   a. While POLQA provides valuable post-call analysis, it is not suited for real-time monitoring of voice quality

Conclusion on Voice Quality Testing

Voice quality testing is an essential aspect of ensuring high-quality communication in telecommunication systems, including IVR systems, call centers, VoIP services, and mobile networks. As voice communication is a fundamental part of many business and consumer services, the need for clear, intelligible, and reliable audio has never been greater. Testing voice quality ensures that these systems deliver the best possible experience for users while maintaining system reliability and performance.

Key Takeaways:

1. **Importance of Voice Quality:**
   a. High voice quality directly influences customer satisfaction, operational efficiency, and service reliability. Poor audio quality, such as distortion, background noise, or delays, can frustrate users, leading to increased call drop rates, miscommunications, and a negative perception of the service.
   b. In sectors like customer support, healthcare, and banking, where clear communication is vital, voice quality becomes even more critical.
2. **Objective vs. Subjective Testing:**
   a. Subjective testing (such as human listeners rating the quality of a call) provides a direct reflection of user experience, but it can be time-consuming and inconsistent.[6]
   b. Objective testing tools like PESQ and POLQA allow for consistent, automated measurements of voice quality, ensuring that systems can be evaluated at scale across different environments and technologies. These tools assess voice quality based on human auditory perception models, providing reliable Mean Opinion Scores (MOS).
3. **PESQ vs. POLQA:**
   a. PESQ has been widely used for many years but is more suited for narrowband systems and may not accurately capture the full range of voice quality in wideband or HD voice applications.
   b. POLQA, the successor to PESQ, is specifically designed for modern communication systems that use wideband, super-wideband, and full-band codecs. It supports higher-quality audio signals (like those in VoLTE, VoWiFi, and HD Voice) and offers improved accuracy in measuring speech clarity, delay, echo, and artifacts caused by network conditions.
4. **Comprehensive Testing is Crucial:**
   a. Comprehensive voice quality testing should cover a range of factors, including clarity, intelligibility, background noise, echo, delay, and packet loss. It should simulate real-world conditions to identify potential issues that could arise during a live call or interaction.
   b. Testing should be done at different levels: from individual calls to network-wide monitoring and benchmarking across different speech codecs, devices, and network types.
5. **Real-World Application:**
   a. Telecom operators, IVR providers, and VoIP services can use voice quality testing to ensure that their services meet industry standards, optimize their systems, and troubleshoot issues in real-time. With tools like PESQ and POLQA, businesses can proactively address quality degradation and ensure a smooth experience for users.
6. **Future-Proofing:**
   a. As voice technology continues to evolve with the rise of 5G, AI-driven speech recognition, and high-definition audio, voice quality testing tools like POLQA will be essential in assessing the increasingly complex systems. These tools ensure that the communication systems are capable of handling higher bandwidth, higher fidelity, and more advanced voice features, all while delivering an optimal user experience.

Final Thoughts:

Voice quality testing is a critical process for any system that relies on spoken communication. Whether it's a customer service call center, a VoIP solution, or a cutting-edge AI-powered IVR, ensuring that the voice quality meets user expectations is crucial for maintaining customer satisfaction and brand trust. By utilizing advanced tools like PESQ and POLQA, companies can objectively and accurately assess voice quality, identify potential issues, and optimize their systems for superior performance. Ultimately, voice quality testing helps create a more seamless, efficient, and enjoyable experience for the end user.

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