Voice over Internet Protocol (VoIP) is being widely deployed to offer users low-cost alternatives for long-distance and international telephone calls. However, for users who are accustomed to the high-quality sound offered by traditional Public Switched Telephone Network (PSTN) service, VoIP voice quality often pales in comparison. Voice quality in VoIP networks widely varies as a result of several factors. This article will discuss how service providers can avoid service degradations in a VoIP network and deliver voice quality that is comparable to, or even exceeds, PSTN levels.
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Overview of PSTN

Voice quality in PSTN is intelligible, sounds natural, enables users to identify speakers and has minor disturbing impairments. Here, voice is delivered as an analog signal, which is terminated at the nearest Digital Loop Carrier (DLC) or Central Office (CO). The DLC converts the analog signal into digital samples for long distance communication, and then at the termination point, another DLC converts the digital samples back into analog form on the TIP-RING telephone interface. High-quality voice is achieved because analog voice signals are sampled at 8000 Hz and compressed to only 8-bits per sample using ITU-T-G711 Pulse Code Modulation.

Factors that Maintain Voice Quality

Several factors that enable the PSTN to maintain voice quality include:

- **Samples and Voice Compression**: Transmissions from the originating DLC to the destination DLC is digital. Digital transmissions with G711 are controlled with stratum clocks, ensuring synchronous sample delivery. Compressed sample miss/slips rarely occur and, as a result, are not noticed by listeners.

- **Delays**: End-to-End delays result from physical transmission delays in the PSTN. There is no memory and packet processing in the transmission, which makes the interactive voice conversation more natural and comfortable.

- **Echo**: Even though the same analog phones are used in PSTN and VoIP calls, echo in PSTN service is not perceivable due to minimal delays. Balanced End-to-End losses incorporated at PSTN DLC and CO help reduce the echo disturbances. For international and long distance calls where delays may be introduced, the carrier grade echo cancellers are incorporated at the long distance termination nodes.

- **TR-57 Compliance**: TR-57 [1] is the North American standard for DLC transmission and switching guidelines. DLC systems and analog front-ends driving the TIP-RING telephone wires comply with TR-57 and local country-equivalent transmission characteristics, which optimize transmission quality. The DLCs in combination with PSTN COs ensure low delays, this results in better perception of call initiation and terminations, as well as use of the call feature services.

- **Ring Equivalent Number (REN) Drive and Impedance Matching**: DLCs are usually designed to support three to five phones on the same physical TIP-RING wires. This enables the PSTN to keep the voice quality despite simultaneous phone usage and minimize the disturbance due to impedance mismatch.

Limitations of the PSTN

The PSTN is not completely fool proof. With some international calls, voice quality can be impaired for a number of reasons. The impairs could be a result of improper terminations, satellite link delays at several G711 trans-coding stages, lack of proper echo cancellers at terminating nodes, long local loops, use of several mismatched impedance phones and use of intermediate packet-based connectivity between long distance nodes.
Why VoIP?

When compared to PSTN, perceived VoIP-based voice quality is influenced by several factors. These factors include delay, echo, voice compression, packet loss, G711-PLC, trans-coding, gateway loss plan and Terminal Coupling Loss – are classified in the TIA-TSB 116-A [2] standard.

Overview of VoIP

In VoIP architecture, the customer premises equipment (CPE) hosts voice services much like the PSTN DLC. The VoIP CPEs also include some functionalities of the PSTN CO, such as call progress tones generation, dialed digits detection, initiating call establishment and supporting call features. VoIP voice is delivered as packets across an IP network. The Figure 1 depicts the several parameters that control voice quality and how the IP network introduces many impediments, such as packet drop, jitter, packet errors and fragmentation.

VoIP service is delivered through CPE such as traditional VoIP adapters, high-end residential gateways and purpose-built IP Phones. The devices rely on local area network (LAN) interfaces, such as Ethernet and wireless LAN (WLAN) or wide area networking (WAN) through digital subscriber line (DSL). The bandwidth available on these network interfaces, the device architecture and the incorporated QoS mechanisms combine to lower the end-to-end packet delays. Many CPE devices manage upstream QoS, while the downstream QoS is managed by the Internet Service Provider (ISP). Data path COs, such as the VDSL Digital Subscriber Line Access Multiplexers (DSLAMs) are now available to support QoS, it eliminates VoIP packet drops at IP terminations.

Trade-Offs and Factors that Influence Voice Quality

The VoIP CPE is designed with certain hardware trade-offs. These trade-offs are used in reducing the cost of the devices in order to make service affordable to the consumers. For instance, clock references are not able to match with PSTN stratum clocks and some of the low-
cost front-end devices used in CPE do not fully meet TR-57 specifications. However, recently several manufacturers introduced low cost telephone interfacing devices that incorporate TR-57 characteristics, which enable better voice quality. Nevertheless, for multiple country impedances, tones, call feature deviations, and loss planning matching to local loudness ratings must be incorporated for overall improved voice quality.

In some countries, Internet and VoIP services are provided by different service providers. To reduce expenses, subscribers may select lower bandwidth Internet service. To overcome these bandwidth limitations, VoIP providers use codecs like G729A instead of G711. The G729A voice compression is eight times greater than G711, thus resulting in a lower quality sound. Additionally, VoIP calls may go through several gateways and trans-coding or conversions from one compression scheme to another in order to reach the end users of other VoIP networks, PSTN or wireless networks.

Dropped packets are unavoidable due to network congestion. Packet Loss Concealment (PLC) algorithms are incorporated into CPE to manage unexpected packet drops. During silent periods, jitter buffers are adjusted to optimize available packets from the network interface and minimize buffering delays. This activity also improves packet impediments that result from end-to-end clock drifts.

Echo is one of the critical items that influence the voice quality. End-to-end VoIP call delays and loudness ratings also contribute to echo. Less delay helps ameliorate echoes. Delays can be reduced with greater bandwidth, QoS mechanisms and CPEs designed for lower packet processing delays. Loudness rating is decided mainly by the end user phones, impedances, end-to-end losses and country of deployment. VoIP voice quality is further enhanced by incorporating carrier grade echo cancellers in the CPE processing.

In VoIP networks, voice quality must be monitored continuously. Voice Quality monitoring (VQmon) software is used in several deployments, but not all CPE devices can support dynamic quality improvement based on the monitored parameters. To achieve the highest voice quality, CPE devices and end-to-end deployment infrastructure must be capable of supporting dynamic quality improvement operation based on the monitored parameters.

Improvements

As this article had detailed, a number of improvements are being made to ensure that users experience voice quality comparable to PSTN when they are using VoIP services. However, it is possible that VoIP quality can exceed PSTN levels when using wideband voice. In traditional PSTN and VoIP services, telephone acoustics and processing limit the voice frequencies from 300 to 3400Hz. In wideband voice, acoustic interfaces and processing supports frequencies from 50 to 7000Hz, which supports natural-sounding conversation with a sense of presence.

Wideband compressed voice with G722 codec takes the same bandwidth as G711. Other wideband codecs like G729EV [3] operate at half the bandwidth to that of G711. The listening perceptions of many wideband codecs often surpass the G711 PSTN quality. The voice quality rating ‘R’ of wideband voice is about 20 percent better than the G711 based on wideband R-model, updates given in ITU-T-G107 and G113 recommendations [4,5].

Wideband voice is already being deployed in Europe, supported by wideband DECT (Digital Enhanced Cordless Telecommunications) phones. These deployments rely on wideband front-end hardware and acoustics. Telephone front-end interfacing Subscriber Line Interface Circuit and sampling hardware devices support wide bandwidth. Currently, several conference phone models, soft phones on personal computers, and some of the IP phones support wideband voice. The acoustic interfaces on digital, Bluetooth, DECT, Wi-Fi phones also can be used for wideband voice services.

A number of improvements are being made to ensure voice quality is comparable to the PSTN when using VoIP services, and also that VoIP quality can exceed PSTN levels when using wideband codecs. In traditional PSTN and VoIP services, telephone acoustics and processing...
limit the voice frequencies from 300 to 3400 Hz. In wideband voice, acoustic interfaces and processing supports frequencies from 50 to 7000 Hz, which provides natural-sounding conversation with the sense of presence. Wideband compressed voice takes less bandwidth than G711.

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