3GPP and 3GPP2 Packet-Based Voice Transport Technologies and Packet-Based Voice Quality Enhancement Technology

Introduction

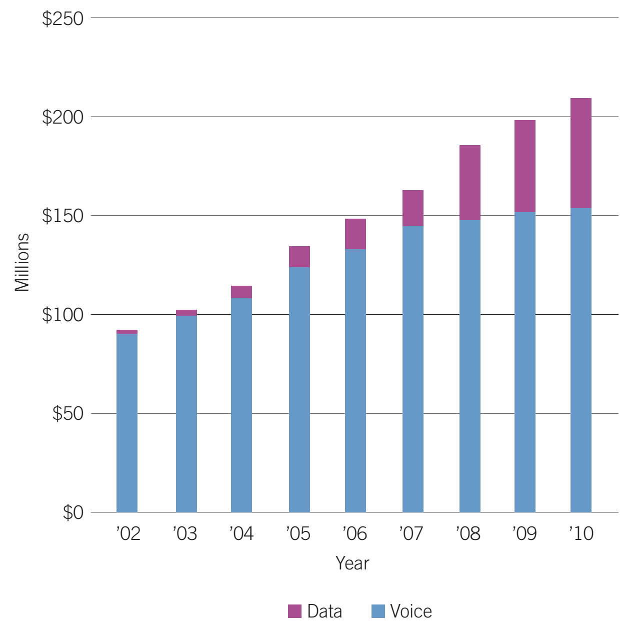
"Mr. Watson! Come here, I want to see you."

Since Alexander Graham Bell uttered those words to his assistant during the first telephone call on March 10, 1876, voice telecommunications has continued to evolve; from party lines to Bluetooth, from wireline to wireless. Even with the popularity of e-mail, text messaging and instant messaging, voice communications is still the dominate form of inter-personal communication. With convenient and ubiquitous coverage and service, wireless communication has become an integral part of our daily lives. And as the wireless market grows, and as competition to increase market share increases, a high level of Voice Quality remains as an integral requirement for these networks. This white paper discusses voice transport technological evolutions in 3GPP[1] and 3GPP2[2], the causes and the types of voice impairments, and the voice quality enhancement technologies treating those voice impairments.

Voice is Still the King

The first fully automatic mobile phone system, Mobile Telephone System A (MTA), was introduced in 1956. Wireless technology has developed from the early, first generation and second generation (2G) technologies to today's 3G and 4G services. Wireless data communications were very limited in the early days due to the lack of speed and practical applications. But, with the introduction of CDMA2000 1XEV-DO and HSDPA/HSUPA, wireless data applications have become a reality. Instead of the bulky, cumbersome "Brick" phones of yesteryear, we now use compact, sleek, mobile phones with music, data and video capabilities settling easily in the pockets of our jeans. Despite these changes, one thing remains the same — voice is still the king "Killer Application" in wireless. Voice communication is still the major revenue contributor for WSPs not only in North America, but globally as well. Figure 1 illustrates the voice and data revenue for the North American market.

High voice quality is one of the primary requirements when consumers choose a WSP. The importance of voice quality as a significant factor in WSP selection is outlined in survey data from such sources as Consumer Reports[3] and the J.D. Power and Associates "Wireless Call Quality Performance Study℠."



Source: Gartner 2007
Figure 1. North American mobile service revenues.

[4] WSPs use this information to their advantage and WSPs with high voice quality numbers site their voice quality in their marketing and advertising campaigns.

High voice quality attracts and retains high revenue customers while the converse is true for low voice quality. Low voice quality leads to higher churn rates. The relationship between churn rates of the major WSPs and their voice quality rankings is clear to even the unsophisticated observer.

Voice Transport, a Historical Perspective

So how did we get to a high speed, data capable network from the early beginnings of wireless communications as a voice only means of communications? The ITU-T G.711 Pulse-Code Modulation (PCM) standard has been used worldwide for voice transmission.

[1] *Third Generation Partnership Project is a global standards body involved with GSM and UMTS*

[2] *Third Generation Partnership Project 2 is a global standards body involved with CDMA*

[3] *ConsumerReports.com, September 2007, URL: http://www.consumerreports.org/cro/electronics-computers/phones-mobile-devices/phones/cell-phone-service-providers/cell-phone-serviceE1-07/overview/0107_serve_ov_1.htm*

[4] *http://www.jdpower.com/telecom/ratings/wireless/call_quality/index.asp*

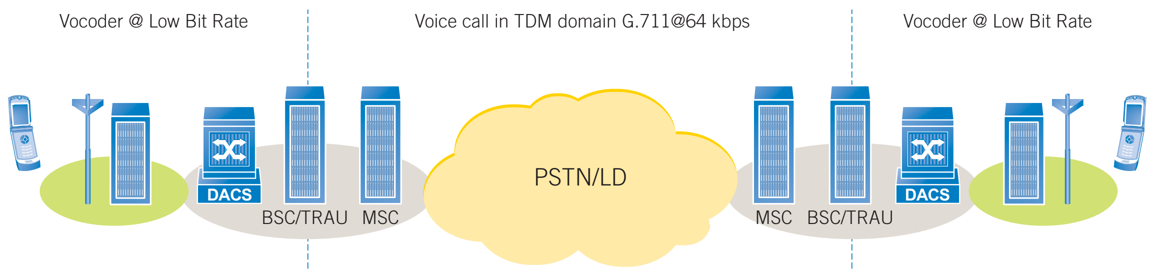


Figure 2. Generalized view of vocoder function.

This standard encodes voice using a 64 kbps bitstream for the voice signal sampled at 8 kHz. It was soon evident that 64 Kbps bits per call would consume the available Radio Frequency (RF) spectrum (bandwidth). Consequently, more bandwidth efficient wireless voice encoders/decoders (codecs), or vocoders, were developed to reduce the bit rate and spectrum usage, enabling more calls to traverse the scarce resource of radio spectrum bandwidth. Calls entering the originating Mobile Switching Center (MSC) are transcoded to the G.711 format at 64 kbps per call. The calls are then transported at 64 kbps per call to the destination MSC or Class 5 Switch for landline calls. Mobile-to-mobile calls will be transcoded from G.711 to the lower-bit-rate codec at the destination MSC per the destination mobile vocoder specification.

As shown in Figure 2, when a lower bit rate voice call enters the Base Station Controller (BSC), Transcoding Rate Adaptation Unit (TRAU) or Mobile Switching Center (MSC), the voice call is transcoded into G.711 64 kbps PCM signal. The call is then transported via the Public Switched Telephone Network (PSTN) or Long Distance (LD) network to reach the called BSC, TRAU or MSC. If the called number belongs to a mobile station, the G.711 voice is transcoded back to the called MS vocoder. The actual transcoding device location varies depending upon the network standards (e.g. 3GPP vs. 3GPP2), as well as on the individual infrastructure supplier.

In an M-M call, transcoding activities at both ends appears to be an unnecessary step. It not only adds substantially to the end-to-end delay, each time the voice is transcoded, it degrades voice quality. In addition, using a low bit rate encoder wastes precious backbone transport bandwidth. Removing the superfluous transcoding activity should have been done long ago. But it — prior to today — it has been easier said than done. The requirements of circuit-switched technology and the signaling complexity required to bypass the Transcoding operation have been formidable obstacles to Transcoder Free Operations (TrFO).

Voice Network Evolution and Benefits

New technologies have been developed for mobile-to-mobile voice calls to improve spectrum and bandwidth utilization while improving voice quality. The more important developments, which we will focus on in this white paper, are 3GPP2 Transcoder Free Operation (TrFO)[5] for CDMA networks, 3GPP Tandem Free Operation (TFO) and Transcoder Free Operation (TrFO) for GSM and UMTS networks. The implementation of these technologies varies significantly between the standard bodies and wireless infrastructure vendors and their carrier customers. However, they all share the same basic principle of transporting the native-coded wireless voice traffic across the network at lower bit rates when compared to the traditional 64 kbps transmission standard. It should be noted that TrFO and TFO were originally designed to support circuit-switched mobile station voice communications, not packet-based voice services, such as Voice over Internet Protocol (VoIP) with a Session Initiation Protocol (SIP)-based mobile.

There are three major categories where these new technologies deliver their benefits:

- Reduced Operation Expenses (OpEx) in three key areas:
  1) Reduced bandwidth requirements throughout the network — more efficient transport lowers total transport costs
  2) Reducing the burden on the switch to perform the Transcoding operations frees up cycles on the switch to process other calls
  3) Eliminating the vocoders reduces the royalties paid for the vocoders
- Reduced end-to-end delay: since the native coded voice stream is used throughout the mobile-to-mobile call, it is not necessary to transcode the native coded voice to G.711 in the originating MSC, nor to transcode the G.711 back to the native coded voice in the destination MSC. The elimination of these two transcoding functions reduces end-to-end delay.

[5] Remote Transcoder Operation (RTO) is also standardized by 3GPP2, but covers mobile to landline applications.

- Improved Voice Quality: Transcoding involves sampling, which inherently reduces the true voice signal, eliminating two transcodings, coupled with reduced transport delay delivers a positive impact to the total voice quality of the call.

As a result of these benefits, WSPs are eager to adapt TrFO and TFO. This transition, however, has taken different paths due to the architectural and infrastructure implementation differences and limitations.

In the following sections, we discuss the different wireless network standards and how TrFO is implemented.

Wireless, a Divided World

As mentioned, there are two major mobile/wireless standard bodies: 3GPP and 3GPP2. Each group designs and implements its architecture quite differently. In the following sections, we will discuss how TrFO is implemented in both 3GPP (GSM/UMTS) and 3GPP2 (CDMA).

Mobile-to-Mobile Voice Calls in a GSM/UMTS Network

The original narrowband Adaptive Multi-Rate (AMR) vocoder was standardized by 3GPP in October 1998 and has been widely used in GSM networks worldwide. AMR-Narrowband (AMR-NB) consists of eight codec modes with different source bit rates, from 12.2 kbps down to 4.75 kbps, and is compatible with the traditional PSTN audio bandwidth at 100 to 3,500 Hz. To enhance the voice quality and better utilize the air interface resources in the UMTS networks, 3GPP standardized the AMR-Wideband (AMR-WB) codec in March 2001. Later, ITU-T adopted AMR-WB as Recommendation G.722.2. The ITU-T adoption was a significant milestone as it was the first time both wireless and wireline networks used the same codec for voice. AMR-WB contains nine different codec modes with source bit rates from 6.6 kpbs up to 23.85 kpbs and expands the audio bandwidth to 50 to 7000 Hz, providing superior voice quality when compared to AMR-NB.

Figure 3 demonstrates how a mobile-to-mobile voice call is transported through the legacy network. Note that the coded signal in the air includes error correction. Sometimes, Transcoder (TC) is used in place of TRAU in this type of architecture diagram.

To gain the OpEx savings, the reduction in voice path delay and the other benefits outline above, a GSM-based WSP has two choices: Tandem Free Operation (TFO) and eTFO (enhanced TFO). These two options are discussed in the following sections.

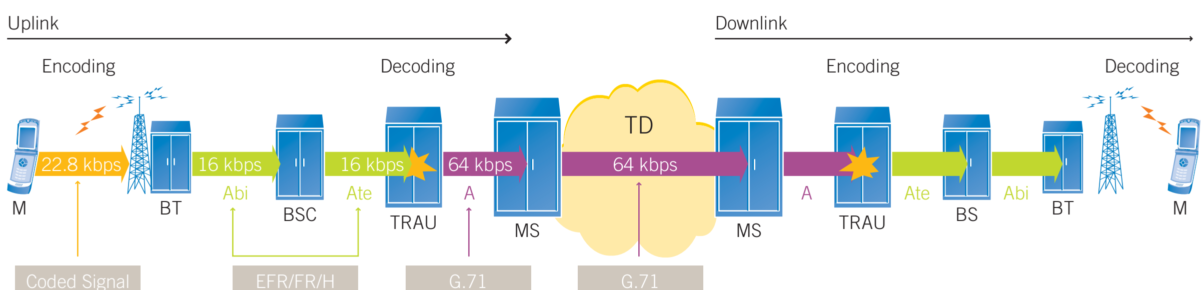


Figure 3. How a mobile-to-mobile voice call is transported through the network.

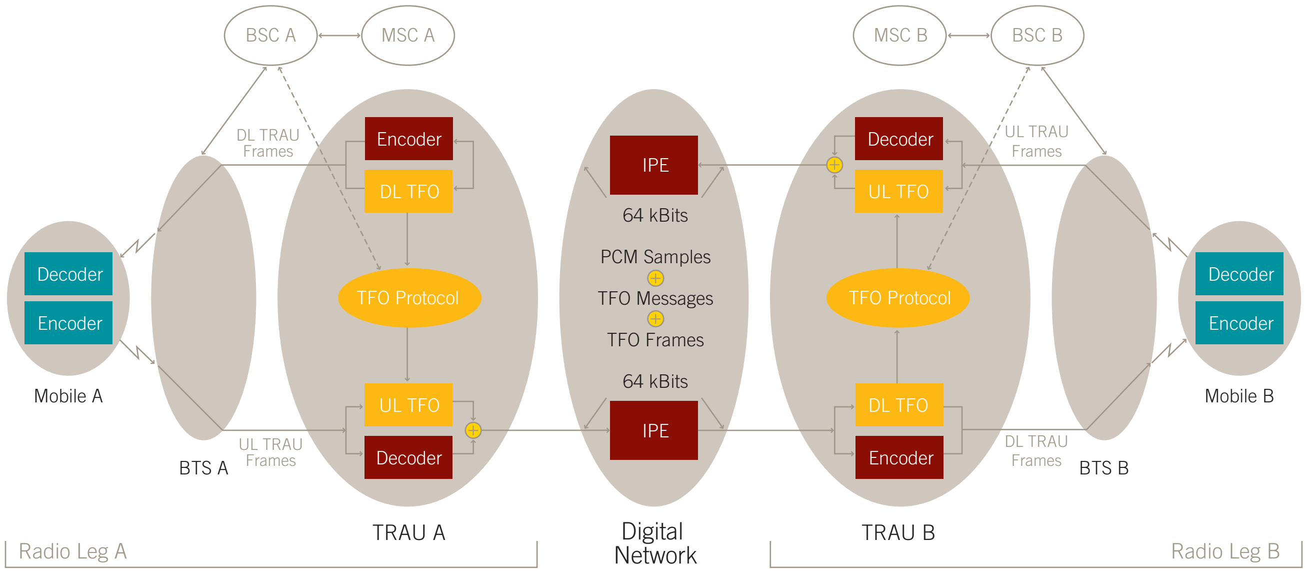


Figure 4. Logical view of TFO operational elements.

TFO in GSM Network

In an attempt to improve the voice quality in GSM networks, TFO (ETSI 101 504; 3GPP TS 28.062) is a viable option. Due to the architecture limitation of the GSM A-interface, which is always G.711 Pulse Code Modulation (PCM) over a 64 Kbps circuit, TFO uses a special bit pattern as part of the G.711 voice stream to negotiate the codec type during a call set-up. TFO utilizes in-band signaling to accomplish this negotiation which, if successful, bypasses the transcoders at both ends and establishes a TFO call (e.g., an AMR at full rate or FR_AMR call) by using a tunnel with the G.711, 64 Kbps circuit. Note that there are no bandwidth savings in transporting the TFO call. However, voice quality is improved due to the elimination of the two previously required transcoding operations. In addition, extra network elements are needed to accomplish an FR-AMR call. Figure 4 displays the logical view of TFO operational elements.

The TFO protocol is the heart of this operation, and negotiates with the called TRAU for compatible codec type using in-band signaling. In-Pass Equipment (IPE) is used to monitor the TFO traffic in the core network. However, when the GSM service provider adopts UMTS with Release 4 (ATM) and/or Release 5 (IP) architectures, the TFO bandwidth savings can be achieved in the core using ATM or IP transport.

eTFO in GSM Network

Enhanced TFO (eTFO) is similar to TFO and enables a GSM network to achieve bandwidth savings in the core when ATM transport is used. eTFO is fully backward-compatible to TFO. Voice calls transported in the ATM network are classified as Real-Time Variable Bit Rate (RT-VBR) traffic. Bandwidth savings are realized via multiplexing RT-VBR packets.

TrFO in UMTS Network

The UMTS core network (Release 4 and up) has a layered architecture with Bearer Independent Call Control (BICC) where the actual bearer data transport path is transparent to the signaling path (Figure 6).

The BICC is also sometimes referred to as Bearer Independent Core Network (BICN).

TrFO uses Out-of-Band Transcoder Control (OoBTC) in signaling to negotiate the voice codecs in mobile-to-mobile call scenarios. The MSC servers will establish a bearer path without activating the voice transcoders in the media gateways between the two mobile phones. Since most UMTS operators are likely on Release 4 with ATM transport or evolving to Release 5 with IP transport technology, backbone bandwidth savings for voice can be achieved, in addition to improved voice quality.

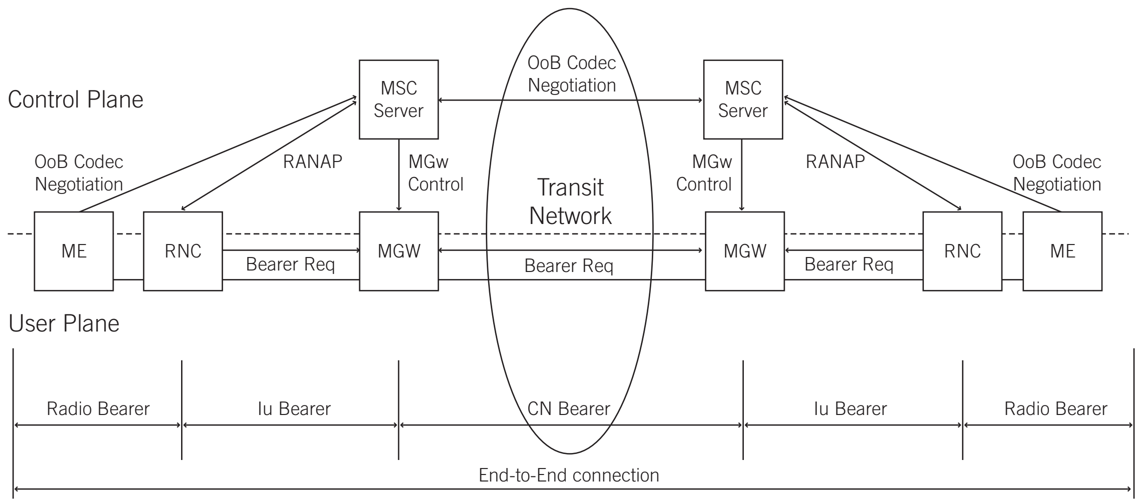


Figure 5. UMTS TrFO architecture (Release 4 and up).

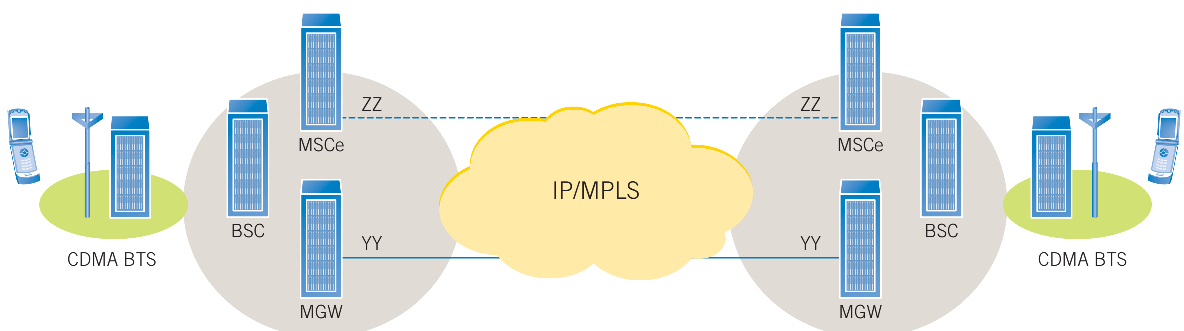


Figure 6. Current TrFO reference architecture.

For a mobile-to-wireline call, the connection remains transcoder-free up to the media gateway, which connects the wireless network to the PSTN. In this scenario, also known as Remote Transcoder Operation (RTO) in 3GPP2, the transcoder is used to convert the AMR-coded voice to G.711 PCM. The voice traffic between the two media gateways remains at the coded AMR domain. In UMTS Release 5, the AMR-WB codec is introduced to leverage the IP network and its higher air-interface bandwidth.

In summary, GSM TFO uses in-band signaling to negotiate codecs at both ends and AMR-coded voice is tunneled with a 64 kbps TDM pipe. Voice quality is improved, but there are no bandwidth savings. UMTS TrFO uses out-of-band signaling to negotiate codecs at both ends and if the UMTS service providers have deployed Releases 4 or 5, both voice quality improvement and bandwidth savings are achieved.

Note that the voice quality improvement achieved by TFO or TrFO is based upon the fact that some finite delays can be removed from the end-to-end voice path due to Transcoding as well as by eliminating the vocoder sampling distortion. These changes in technology employed to transport voice traffic do not address the more prominent sources of voice impairments. These impairments are non-network based and include impairments from the surrounding environment (background noise), user behavior, handset equipment (acoustic echo) and varying RF conditions. These non-network based impairments can only be addressed with Voice Quality Enhancement equipment.

TrFO in CDMA Network

In North America, CDMA infrastructure vendors implement their networks with different physical architectures for historical reasons and to accommodate 3GPP2 standards. Thus, dealing with CDMA networks requires specific knowledge on the network architectural variations, regardless of it being IS-95A/B (CDMAOne) or CDMA2000-1X. The most current TrFO/RTO is defined in 3GPP2 X.S0025-0, Legacy MS Domain Step 2 and a high-level reference architecture as shown in Figure 6.

Based upon this reference architecture and existing network architectures, CDMA infrastructure vendors implement TrFO quite differently in terms of physical network elements and interfaces.

Voice Quality Impairments

Voice quality is impacted by both network related issues and environmental related issues. **Network**-related voice quality issues are hybrid echo, level mismatches, voice intelligibility and delay. **Environmental**-related issues are acoustic echo, background noise and volume levels. Since echo control is the most annoying impairment, we will concentrate on breaking down the sources (causes) and treatments for hybrid and acoustic echo.

Hybrid Echo

Hybrid echo originates at the point where the 4-wire to 2-wire conversion in PSTN networks takes place. Hybrid echo is sometimes referred as electrical echo, because it stems from the impedance mismatch present in the 2-to-4 wire conversion (analog to digital). Hybrid echo will not exist in the mobile-to-mobile calls within a TRFO network because there is no conversion from 2 wire to 4 wire. Hybrid echo, however, will continue to be present in the mobile-to-PSTN calls.

Acoustic Echo

Acoustic echo is caused by acoustic coupling between the speaker and microphone. Acoustic echo occurs in both wireline and wireless networks. The degree of acoustic coupling is dependent on a number of factors. Three prominent factors or sources of acoustic echo are:

1. The minimum displacement between the speaker and microphone
2. The mechanical (physical) design of the handset
3. The environment where the callers are using their handsets

The two main characteristics or measurements of acoustic echo are:

1. Delay
2. The Weighted Acoustic Echo Path Loss (WAEPL)

In some cases, Echo Return loss (ERL) is used instead of WAEPL. WAEPL or ERL is used to determine the intensity or volume of the acoustic echo.

The impact of acoustic echo is dependent on two independent variables:

1. The intensity, volume or loudness of the echo
2. The delay between the original voice signal and the echo

As the delay extends or gets longer, the more noticeable and annoying the acoustic echo becomes to the caller. Similarly, louder echoes are more objectionable than lower volume echoes. Because wireless networks generally impose longer delays compared to wireline networks, the effects of acoustic echoes are more prevalent and noticeable.

Acoustic echo can be caused by some or all of the following:

- Mobile handset design: Early analog mobile handsets were designed like wireline handsets, with sufficient physical distance between the earpiece and mouthpiece. This setup limited the amount of signal passing between them — Terminal Coupling Loss (TCL) — without the need for special echo control circuits. New, smaller mobile handsets have a decreased TCL value and a corresponding reduction in voice quality.
- Headphones and hands-free kits: Hands-free safety laws are becoming more common globally, spurring the proliferation of Bluetooth® headphones and hands-free car kits, both of which have a very low TCL value. Bluetooth encoding and decoding also results in additional delay.

Background Noise

Background noise is an environmentally caused impairment. With the increasing use of cell phones in our everyday lives, these handy devices are being utilized in all types of environments. Calls are routinely placed in locations such as airports, restaurants, shopping malls, inside automobiles or in other high noise environments. The planned networks migration to TrFO will NOT eliminate these impairments, and they will continue to exist and trouble the WSP's customers.

Volume Levels

The many and varied network elements comprising a wireless network and the associated transport network are manufactured to very loose specifications with respect to transmit levels. Many of these elements can affect the volume levels of speech, which will be propagated throughout the network and to the receiver of the call. Tellabs network audits, performed at the request of our customers, have shown transmission levels for wireless calls/networks at a higher speech level than wireline calls. These speech levels can lead to quality issues within customer's calls. As speech levels increase and decrease based on network conditions, it can and will impact the voice intelligibility of wireless calls.

Voice Impairment Scenario	Solution Requirement and Tellabs Feature
High noise levels making it difficult for the other party to hear	Noise reduction algorithm that can detect background noise and remove it to improve call intelligibility. The solution should provide fast and stable convergence with no modulation of the amount of noise reduction provided during active speech periods, nor any adverse impact on quality of the mobile voice signal. Tellabs Feature: Tellabs Noise Reduction (TNR)
Poor signal-to-noise ratio (SNR) for the mobile subscriber	Adaptive Gain algorithm that can enhance downlink speech signals heard by the mobile subscriber when high background noise is present. Signal levels should be adaptively boosted and spectral content be enhanced to improve intelligibility of speech in such situations. Tellabs Feature: Tellabs Adaptive Gain (TAG)
Acoustic/Reflective echo heard by either party on a wireless call	A bi-directional acoustic echo control solution should provide protection from acoustic echo generated in wireless handsets and hands-free kits, minimizing echoes while still providing clear double-talk performance and background audio transparency. This solution must be implemented bi-directionally to treat potential acoustic echoes in either call path. Additionally, an intelligent switching logic is needed to detect the need for either acoustic or hybrid cancellation in the downlink path of wireless calls and introduce the appropriate type of cancellation technology.Tellabs Feature: Tellabs Acoustic Control (TAC)
Hybrid Echo heard by the wireless subscriber	The hybrid circuit in a wireless connection, the source of the largest echo reflections, is located in the wireline service provider's network — yet it is the mobile subscriber that hears the echo. The solution must provide sufficiently long end-path delay coverage up to 250ms to remove the hybrid echo. However, hybrid echo will not present in TFO or TrFO call. Tellabs Feature: Hybrid Echo Control
Imbalanced audio levels in either direction on a wireless call	The solution must be able to achieve an optimal volume level for the mobile callers by boosting or attenuating the audio volume level for both mobile users. Tellabs Feature: Tellabs Level Control (TLC)

Voice Quality Enhancement Technology

As outlined in the section before, many factors degrade total voice quality in wireless communications. In order to remain competitive in the wireless market, the voice quality of each network must be comparable to others in the market, and all are inching closer to the wireline network quality. Improving voice quality means addressing each impairment separately. The table below shows how Tellabs Voice Quality Enhancement (VQE) solution addresses each of the impairments. This table also highlights the Tellabs feature specifically addressing each impairment.

A WSP can selectively implement these solutions individually per its network impairment characteristics or implement them all. Sometimes the Hybrid Echo cancellation function is implemented in the BSC/MSC since it is a well-known and well-defined problem.

Current VQE Placement in a Wireless Network

Before implementing TrFO in a circuit-switched wireless infrastructure, all voice calls and associated voice impairments are transcoded to the G.711 format in BSC or MSC after leaving the Radio Access Network (RAN). Since both voice and impairments are present in the TDM domain, all VQE signal processing takes place in

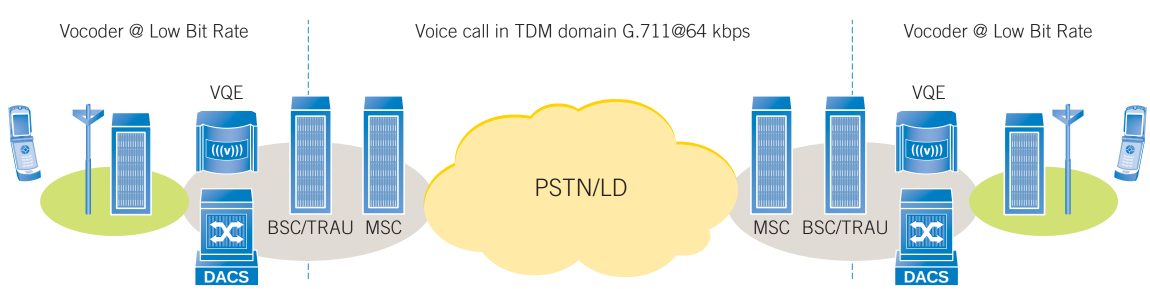


Figure 7. TDM-based VQE equipment in a generic wireless network.

the TDM domain in order to minimize, correct or eliminate voice-related impairments. Tellabs Voice Quality Enhancement (VQE) products not only remove acoustic and hybrid echo, but also automatically suppress background noise and adjust gain, resulting in superior voice quality.

All wireless traffic is in the TDM domain before the introduction of data applications and all interconnections in the Mobile Telephone Switching Office (MTSO) are via the Digital Cross-connect System (DCS). Thus, the TDM-based VQE is typically placed close to the DCS or integrated as part of DCS, as is the case with the Tellabs® 5500 Digital Cross-Connect System Integrated Voice Quality Enhancment (iVQE) product. Figure 7 shows the placement of TDM-based VQE equipment in a generic wireless network.

The Tellabs® 3000 Series VQE product family includes the the Tellabs® 3100 Voice Quality Enhancement System, Tellabs® 3100M Voice Quality Enhancement System, Tellabs® 3300 Voice Quality Enhancement System, and Tellabs® 3600 Voice Quality Enhancement System in addition to the the Tellabs iVQE, which is integrated with the Tellabs® 5500 DCS. As of January 2007, there were more than 300 Tellabs VQE customers in 70 countries worldwide.

Are Voice Impairments Present in TrFO Networks?

Yes, Acoustic echo will be present in TrFO networks. Network audits of TrFO networks performed by Tellabs over the past year have shown the same percentage of acoustic echo issues occurring in both networks. The causes of acoustic echo are not network based, but result from the phones and associated accessories attached to the network as well as the environment the mobile user is calling from. Acoustic echo is network agnostic and is prevalent in all networks.

TrFO Call Capture and Analysis

During the past two years, Tellabs has performed voice quality audits on both traditional and TrFO calls in leading WSP networks across the U.S. The results show that approximately 6% of TrFO calls had objectionable acoustic echo, among other impairments. This is comparable to the frequency of occurrence of acoustic echo in TDM/legacy networks. This is understandable since the causes of acoustic echo are not network dependant.

The TrFO networks experienced a shorter end-path delay (approximately 50 ms less) than seen in studies at similar WSP not utilizing TrFO. Figure 8 shows what levels would be considered objectionable echo at the associated end-path delay. Generally, the louder the echo returned, coupled with the length of delay, determines the acceptability of the acoustic echo. Figure 8 shows the echo objection rate in relationship with TCL and one-way delay per ITU-T G.131 standards.

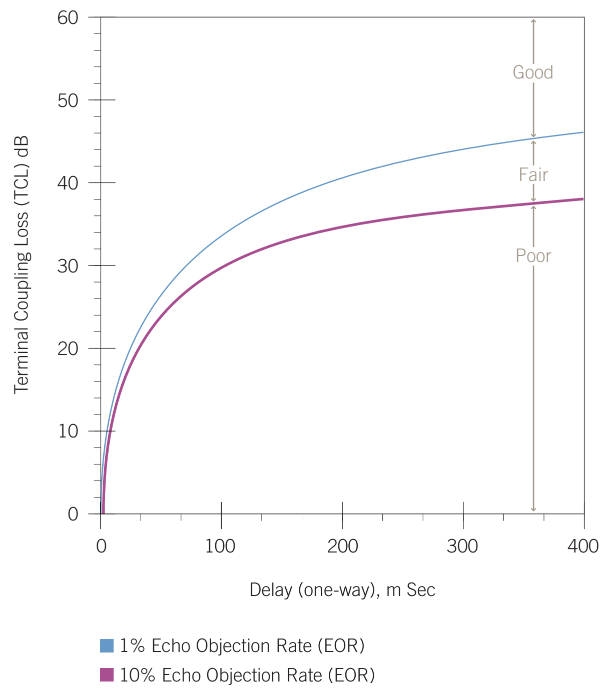


Figure 8: Echo objection rate with respect to TCL and one-way delay.

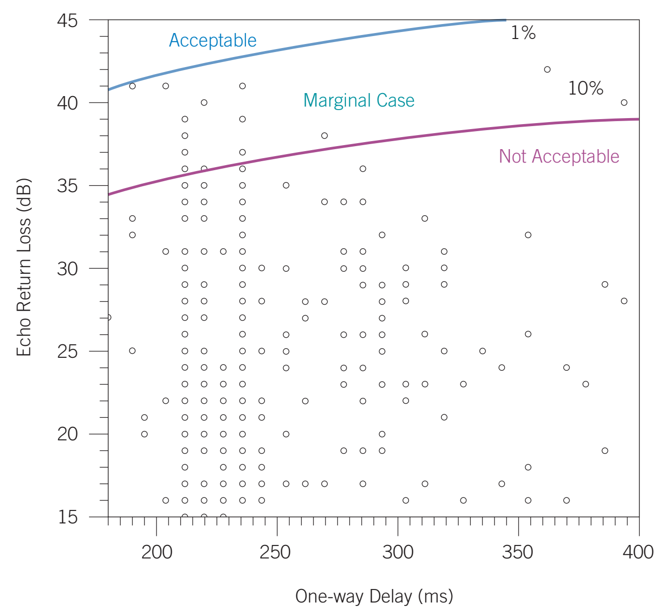


Figure 9: Actual acoustic echo analysis in a TrFO network.

Everything below the pink line in this figure would be considered objectionable acoustic echo resulting in poor call quality reflected by lower PESQ/MOS scores. Figure 8, shows the calls which demonstrated objectionable or marginal acoustic echo for a live TrFO call analysis.

The median network delay for all calls analyzed was 228 ms. In Figure 8, the curve is fairly flat in the 200 to 300 ms range. The conclusion is, as you migrate from the current TDM-based network to a TrFO-based network, the amount of current/existing acoustic echo will remain the same.

Voice Quality Measurement

Mobile voice quality is a subjective measurement of a mobile user's auditory-psychological perception of a call. Many research projects have been performed to objectively quantify the human auditory-psychological response to a call. While many attempts to reflect the human response have been suggested, the Telecomm community has settled on the Mean Opinion Score (MOS) as the index used to describe the voice quality of each call. While the MOS is accepted as the standard to compare one call versus another, there is some discrepancy in the methods used to derive the score. There are three International Telecommunications Union (ITU) standard ways to measure MOS: ITU-T P.563 P.SEAM, ITU-T P.862 Perceived Evaluation of Speech Quality (PESQ) and ITU-T G.107 E-Model.

Mean Opinion Scores

MOS is a set of standards focused on quantifying speech quality in a network. There are two ways of measuring MOS; subjective and objective. Subjective test methods use a number of real human subjects to evaluate the speech quality, either by a conversation or by just listening as the other party speaks. The subjects then rate the call and generally based on a five point scale (1 is poor, 5 is excellent) producing a Mean Opinion Score (MOS). Objective measurement methods use computer-analyzed test signals with perceptual analysis algorithms onboard to produce a predicted MOS score.[6]

Subjective Listening Test

In subjective testing, people judge the quality of real mobile phone conversations. The testing may be either one-way (listening test) or a two-way conversational tests. Subjective testing can be expansive and time consuming. In a one-way test, the user listens to a test signal and subjectively rates the quality of the speech file. This type of testing does not account for acoustic echo, because the person judging the call quality is not experiencing the echo. Two–way testing involves two live people having a conversation across the network. They grade the conversation based on subjective speech quality. Subjective two-way testing is the best way to rate the impact of acoustic echo. However, it is time consuming and expensive.

Objective MOS Testing

Objective testing utilizes testing algorithms to determine on MOS value. A speech file is sent across the network with a known impairment. This is compared to a clean speech file and given an appropriate score. Objective tests do not have the ability to measure acoustic echo. They only operate in a one-way listening mode. The objective tests measure one-way speech distortion and noise quality. They do not measure echo, delay, loudness loss and other impairments related to two-way interactions.

MOS testing is a tool used by operators to help determine the quality of their networks. Subjective tests may be utilized to determine the impact of acoustic echo as well as other environmental/network impairments. Objective tests do not exist that will measure the effect of acoustic echo on the network. Objective tests will help identify the impacts of other impairments such as background noise.

TrFO Implementation

Due to their distinct CDMA infrastructure implementation between CDMA equipment suppliers, TrFO/RTO benefits are realized using various network elements, evolution paths and timelines. The figure below shows a high level architecture diagram on Alcatel-Lucent TrFO Network. The architecture applies to both Intra-MSC and inter-MSC TrFO mobile-to-mobile calls.

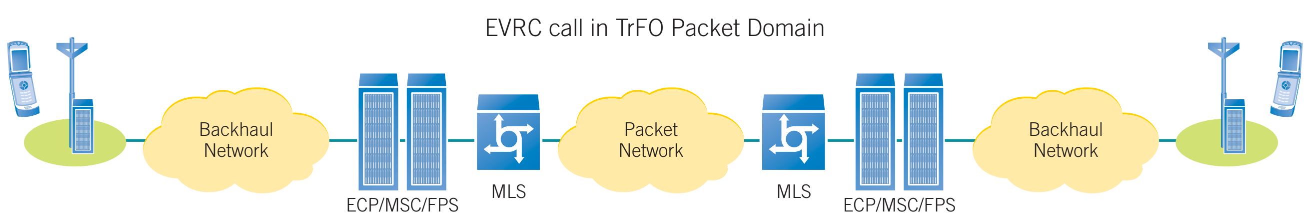


Figure 10: CDMA TrFO network implementation example.

[6] *Predicted MOS scores are produced via a perceptual analysis algorithm aligned close to actual MOS scores derived by mapping the results of subjective scores from databases of speech samples containing defined qualities and impairments.*

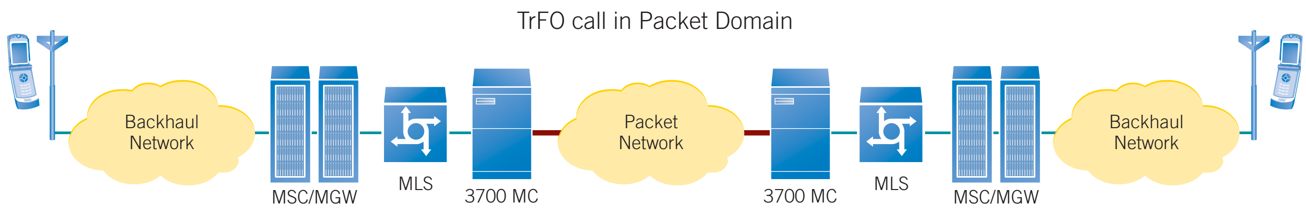


Figure 11: Tellabs® 3700 implementation in a CDMA TrFO network.

Figure 10 illustrates a CDMA TrFO network architecture example, which is implemented by a north American CDMA equipment vendor.

In this scenario, the native, EVRC coded voice is transmitted directly through the network to the called mobile. When the EVRC call enters the ECP/MSC/FPS, the EVRC call is packetized into EVRC/RTP/UDP/IP/Layer2 and is made ready for transport. SIP-based signaling is used to control the call setup and media paths. The role of MLS is to route the packetized calls.

Acoustic Echo Control in a TrFO Network

A straight forward method of performing acoustic echo in a TrFO network is to decode the signal to linear samples, apply traditional linear-domain voice enhancement and then re-encode the enhanced signal into EVRC. However, this approach, in effect, includes a transcoding function since the signal is re-encoded, thereby negating two main advantages of TrFO networks. Therefore, in order to realize the full advantage of TrFO networks, any network-based VQE should not include a re-encode or a transcoding function.

To enhance the voice quality in the native vocoder packetized voice domain, Tellabs has filed various patents in the packet domain VQE area. Based upon the patents and extensive VQE development and deployment experience, the Tellabs® 3700 Multimedia Controller (MC) was developed. The Tellabs 3700 MC will treat the voice impairments in the native vocoder packetized voice domain instead of in the G.711 TDM domain. Figure 11 shows an architectural view how the Tellabs 3700 MC can be implemented.

Conclusion

Regardless the technological differences in the networks and their implementation. To reap the TrFO benefits, all WSPs will pursue it in the near future, depending their network evolution paths and timeline. Thus, it is important to recognize:

1. Acoustic echo is not dependent on the type of network being deployed. With the ever increasing usage of hands-free devices, blackberries, low cost handsets, etc., acoustic echo will continue to be generated in TrFO networks.
2. Transcoding delay has been eliminated in on-net mobile-to-mobile calls. However, in some cases, the Bluetooth headphone and hands-free car kit add additional finite delay.
3. Voice impairments can also be heard more clearly, since less distortion is added due to unnecessary transcoding steps.
4. Voice transport bandwidth savings can be achieved.
5. Packet loss and jitter have greater impact to the coded EVRC traffic as compared to G.711 PCM, since the coded EVRC has lower bit rate.

Therefore, a WSP must differentiate its voice quality amongst its competitors, if the majority of its revenue stream comes from voice. To achieve a superior voice quality in the TDM or packet domain, Tellabs offers a comprehensive VQE product portfolio and extensive implementation experience worldwide. Furthermore, when TrFO is to be implemented, Tellabs is developing and productizing packet-based VQE using its wealth of intellectual properties, patents, and talented and experienced development organization.

Glossary

AMR	Adaptive Multi-Rate (vocoder of GSM)
AMR-NB	AMR (vocoder) — Narrow Band
AMR-WB	AMR (vocoder) — Wide Band
ATM	Asynchronous Transfer Mode
BICC	Bearer Independent Call Control (used in UMTS TrFO)
BSC	Basestation Controller
CDMA	Code Division Multiple Access
CODEC	Compressor-Decompressor or Coder-Decoder
DCS	Digital Cross-connect System
eTFO	enhanced Tandem Free Operation
EV-DO	Evolution — Data Optimized (of CDMA2000-1X)
EVRC	Enhanced Variable Rate CODEC
EVRC-B	Enhanced Variable Rate Codec B (next generation to EVRC)
GSM	Global System for Mobile communications
HSDPA	High-Speed Downlink Packet Access (of UMTS)
HSUPA	High-Speed Uplink Packet Access (of UMTS)
IP	Internet Protocol
LD	Long Distance
ME	Mobile Equipment
MGW	Media Gateway
MLS	Multi-Layer Switch
MS	Mobile Station
MSC	Mobile Switching Center
MTSO	Mobile Telephone Switching Office (that houses the MSC and other equipment)
OoB	Out-of-Band
OoBTC	Out-of-Band Transcoder Control (used in UMTS TrFO)
PCM	Pulse-Code Modulation
PSTN	Public Switched Telephone Network
RAN	Radio Access Network
RANAP	RAN Application Part
RNC	Radio Network Controller
RTO	Remote Transcoder Operation
RTP	Real-time Transport Protocol
TDM	Time Division Multiplexing
TFO	Tandem Free Operation
TRAU	Transcoder and Rate Adaptation Unit
TrFO	Transcoder Free Operation
UDP	User Datagram Protocol
UMTS	Universal Mobile Telecommunications System
VoIP	Voice over IP
VQE	Voice Quality Enhancement
WSP	Wireless Service Provider

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