Journal of the National Academy of Forensic Engineers®

http://www.nafe.org
ISSN: 2379-3252

Vol. 38 No. 1 June 2021

FE Analysis of Communications Systems for Drive-Thru Restaurants in a Business Dispute Over Specifications and Design Process

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Abstract

Forensic analysis in this case involves the design of a communication system intended for use in Quick Service Restaurant (QSR) drive-thru lanes. This paper provides an overview of QSR communication system components and operation and introduces communication systems and channels. This paper provides an overview of non-linear, time-varying system design as contrasted with linear, time-invariant systems and discusses best design practices. It also provides the details of how audio quality was defined and compared for two potentially competing systems. Conclusions include that one of the systems was clearly inferior to the other — mainly due to not following design techniques that were available at the time of the project.

Keywords

Forensic engineering, quick service restaurant, QSR, drive-thru, communications system, simplex, half duplex, full duplex, adaptive systems, automatic gain control, AGC, noise cancelation, linear time invariant, LTI, nonlinear time-varying

Introduction

The plaintiff's complaint refers to the system in question as a Drive-Thru System. The system is intended to provide communication between service employees at quick-service restaurants (QSRs) and their customers. Typically, a customer enters a drive-thru lane outside the QSR and approaches a structure referred to as the post, which includes a microphone and a speaker.

The system detects the customer's presence and alerts the serving employee. The serving employee, using a headset with microphone and speaker, greets the customer. The employee and customer converse through a two-way communication channel. Following the conversation, the customer proceeds to the service window where the transaction is completed.

The buyer (plaintiff) in this case had an established reputation for reselling and repairing QSR communications systems and decided to manufacture them under its own brand. It outsourced the design to the designer (defendant) who claimed expertise in radio design. Expected deliverables included assembled exemplar units, schematics, software, parts-lists, diagrams, and assembly/service instructions. The buyer and the designer agreed to specifications, schedule, and cost. Both cost and schedule were overrun. The buyer demanded contract rescission and refund. The buyer sued the designer after negotiations failed. The author was retained by the buyer's attorney to investigate and opine on:

- Measured audio quality of the system. Did it meet specifications?
- Did the designer follow best design practices?

History and Overview

Red's Giant Hamburg, on Route 66 in Springfield, Illinois, opened the first drive-thru in 1947. Since Red's closed in 1984, it is likely that In-N-Out Burgers, operating since 1948, is now the longest-running fast food restaurant offering a complete drive-thru package. At the time of its opening, In-N-Out's drive-thru system included a state-of-the-art two-way speaker box[1]. Other drive-thru early adopters were Jack-in-the-Box in 1950 and Wendy's in 1969. In the mid-1970s, McDonalds opened its first drive-thru lane[2].

A Quick Service Restaurant is the restaurant industry's term for what people usually call a fast-food restaurant[3].

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DOI: 10.51501/jotnafe.v38i1.106

Most major QSR chains report that drive-thru lanes account for about 70% of sales — and that accuracy and speed of service are the two most critical drive-thru metrics[4].

The technology behind the communication is a key link to drive-thru accuracy. “The quick-service industry relies on efficient communication. A poor sound quality can lead to incorrect orders/delays and can greatly impact the quality of service and customer experience[5].” Taking the order correctly relies on technology. Poor performance of the drive-thru speaker and employee headsets can lead to inefficiencies and customer loss. Digital communication has replaced analog for the communications system for several reasons, which will be developed more fully later.

1. Digital communication systems can be designed to reduce noise and external interference.

2. Digital signal processing can be used to make speech more intelligible.

3. Digital systems can perform non-linear time varying control over voice communication.

Communications System

This section is a general overview of audio communications systems. Simplex, half duplex and full duplex are three types of communication channels[6]. See reference for a full introduction to digital communications systems[7].

Simplex

A one-way communication channel is referred to as a simplex channel. An example of a one-way communication channel is an on-stage announcer speaking into a microphone, with the announcer’s voice coming through a speaker to the audience. A channel in this announcer/audience example is the electronic (or wireless) set of equipment forming a path from the announcer to the audience. Radio and television station broadcasts are other examples of simplex communication.

Half Duplex

Combining two one-way communication channels into a two-way communication channel, with which only one person may speak at time, is referred to as a half duplex channel. A pair of walkie-talkies or a pair of CB radios are examples of half duplex channels. Conversations over half duplex channels may include jargon such as a talker finishing a speech segment by saying “over.” In other examples, there is a button that the talker presses and holds while speaking. It is up to channel users to cooperate and take care to share a half duplex channel appropriately.

Full Duplex

A two-way communication channel, allowing participants to speak simultaneously, is referred to as a full duplex channel. In ordinary face-to-face conversation between two or more people, any participant may speak at any time — reciting together, singing together, interrupting, or talking over a different participant. A full duplex communication channel allows for this ordinary and natural-like communication. When talker-B interrupts talker-A, talker-B continues to hear what talker-A is saying, and talker-A hears the interruption. Talkers resolve the interruption the same way they do while conversing face to face.

The telephone service we have known all our lives is an example of a full duplex system. Maintaining high-quality full duplex operation throughout the evolution from analog wired telephone service first to digital wired telephone service and then to digital wireless telephone service initially required significant effort on the part of telecommunications engineers. Full duplex functionality is now commonplace in both wired and wireless telephone service.

Earlier generations of drive-thru systems used half duplex. However, half duplex conversations may result in miscommunication, and overall they seem unnatural to some people. A customer may interrupt the employee to change an order while the employee is speaking. The employee may not realize the customer’s requested order change, reducing the quality of the system’s functionality and perhaps increasing costs to the QSR.

QSR System Functional Block Diagrams

Component placement throughout QSR systems may differ, but the following components are part of the communication system.

As indicated in **Figure 1**, the microphone converts the sound pressure of the employee’s voice to a voice-band analog electrical signal. The amplifier increases the amplitude of the electrical signal and filters it in preparation for digitizing. The digitizer converts the voice-band analog signal to a stream of digital words (digital signal) and encodes a base-band signal in preparation for modulating an RF carrier. The radio transmitter generates a radio frequency (RF) carrier, modulates it with the base-band digital signal, and transmits it as a wide-band RF signal over the air.

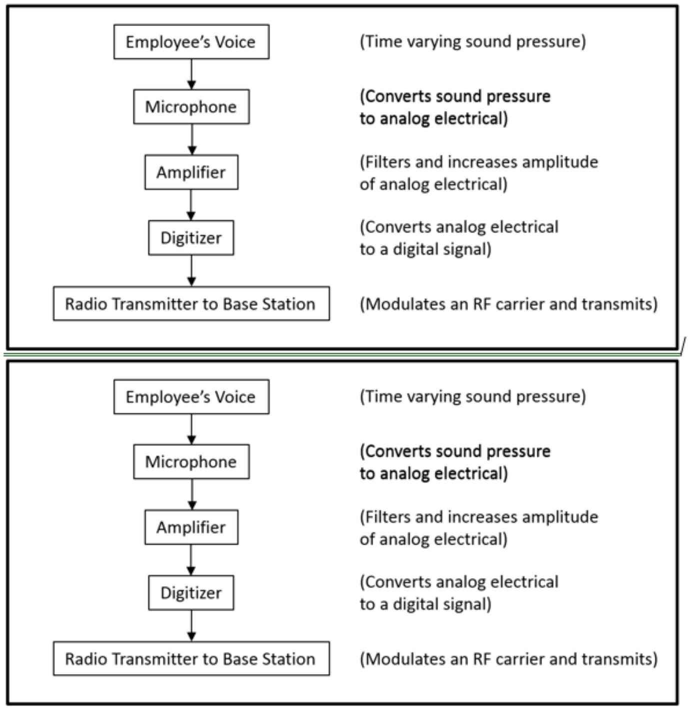

Figure 1
Outbound path – block diagram of employee's headset.

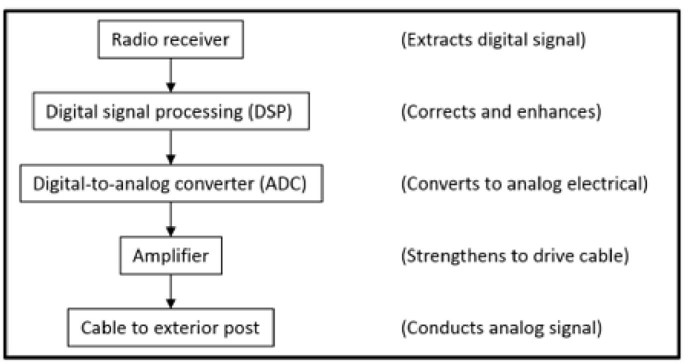

Figure 2
Outbound path – block diagram of base station.

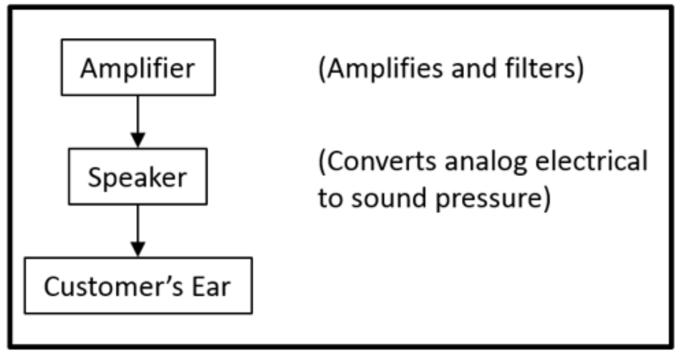

Figure 3
Outbound path — exterior post in drive-thru lane.

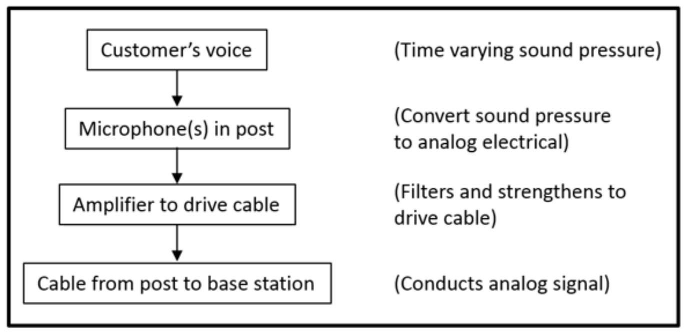

Figure 4
Inbound path — exterior post in drive-thru lane.

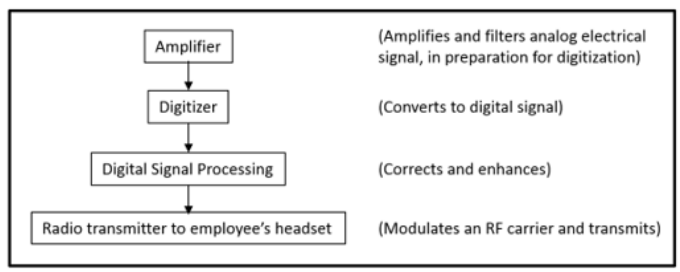

Figure 5
Inbound path — base station.

Outbound Path – Employee's Headset

Outbound Path – Base Station

Typically, the base station is mounted on a wall in the drive-thru booth, within a few meters of the employee. As indicated in **Figure 2**, the base station contains a radio receiver that demodulates the wide-band RF signal from the employee's headset, extracting the digital message signal from the base-band signal. Multi-function DSP, as described later in the paper, corrects and enhances the digital signal. The processed digital signal is then converted to an analog electrical signal, which is driven through a cable to the post.

Outbound Path – Exterior Post in Drive-Thru Lane

As shown in **Figure 3**, the exterior post in the drive-thru lane completes the path from the employee's microphone to the customer's ear, by way of the amplifier and speaker.

Inbound Path – Exterior Post in Drive-Thru Lane

As indicated in **Figure 4**, microphones in the external post convert the sound pressure of the customer's voice to a voice-band analog electrical signal. The amplifier amplifies, filters, and conditions the signal, and drives it through the cable to the base station.

Inbound Path – Base Station

As shown in **Figure 5**, the inbound amplifier in the base station receives the analog electrical signal conducted by cable from the external post. It amplifies and filters the signal to prepare it for digitization. Multi-function DSP corrects and enhances the digital signal, converting it into a base-band signal suitable for modulating an RF carrier. The radio transmitter generates and modulates an RF carrier with the base-band signal and transmits the resulting wide-band signal over the air.

Inbound Path – Employee's Headset

As indicated in **Figure 6**, the employee's headset contains a radio receiver that demodulates the wide-band RF signal from the base station, extracting the digital message signal from the base-band signal. Multi-function DSP corrects and enhances the digital signal. The processed digital signal is then converted to an analog electrical signal, which is amplified and conditioned to match the headset's speaker. The speaker converts the analog electrical signal to sound pressure heard by the employee.

Digital Signal Processing (DSP)

From the electronics point of view, the major differentiator of product quality is its digital signal processing. DSP algorithms enhance the audio environment of the system. DSP features include automatic gain control (AGC), noise reduction, and echo suppression.

Automatic Gain Control (AGC)

AGC works to keep the speaker/headphone sound volume constant. Some customers speak louder than others. The employee needs to understand the customers to fulfill orders accurately and quickly. DSP measures the loudness of customers' conversation and compares the loudness to a desired level set by the employee. When the customer speaks softly, DSP turns up the volume automatically. DSP turns down the volume automatically when customers speak too loudly. This is the essence of AGC, which is used in both the outgoing and incoming paths.

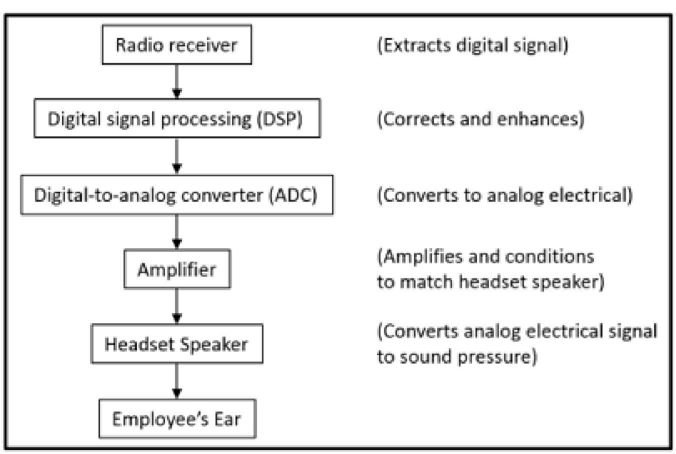

Figure 6
Inbound path — employee's headset.

AGC has been used in communication systems for a long time and was accomplished prior to DSP in totally analog systems[8]. DSP AGC procedures are well known to system designers. See reference for an example[9].

Noise Reduction (NR)

For drive-thru systems, it is convenient to define "noise" as any unwanted sound other than the talker's voice[10]. Strictly speaking from an electrical engineering and physics point of view, this usage of the term "noise" ought to be called "interference plus noise." Typical sources of interference, which the parties to this case call "noise," include smoothie blenders inside the restaurant and automobile engines in the drive-thru lanes. In physics and electronics, noise is random and unpredictable. Electronic noise refers to the thermal, shot, flicker, burst, and transit-time noise, which comprise total physical noise-power, which is compared to signal-power to derive the Signal-to-Noise Ratio (SNR)[11]. Noise reduction techniques for random, unpredictable electronic noise are different than those for interference, such as from blenders and car engines[12].

The "voice" signal from microphones includes not only the voice of the talker, but also the background sounds the microphone picks up. For the inbound path from customer to employee, unwanted noise may include sounds from the customer's car or motorcycle engine, or noise from other customers' vehicles, passing traffic, lawn mowers, machinery, etc. For the outbound path from employee to customer, noise may include sounds made by kitchen machinery, such as mixers, blenders, and others. Certain system design flaws can cause audible humming from the speaker, which is picked up by the microphone. Unwanted system sound is considered noise in this context. All these sounds (along with talker's voice) get amplified and digitized.

DSP noise reduction is relatively new, dating from the 1980s[13]. A typical DSP procedure examines the digitized signal and looks for recurring patterns that share characteristics with the noise sources described in the previous paragraph and goes on to reduce their loudness. For instance, the customer may still recognize the sound of a blender, but when DSP is operating properly, that sound is reduced enough for the customer to understand the employee.

Echo Suppression

Echo in a drive-thru system can refer to talkers hearing their words repeated through the speaker after a delay. Echo suppression in this context can be accomplished with a DSP function called autocorrelation which examines the digitized signal and looks for repeated patterns corresponding to previously processed voice signal[14]. As in a typographical rendering of an echo "HELLO … Hello … hello …," the DSP procedure detects and removes the "Hello … hello …" from the speech signal being processed. Let us refer to this as "double-talk" echo suppression.

Echo suppression may also refer to suppressing the squealing sound (popularly called "feedback") that one hears when a microphone is placed too close to a speaker. This squealing can occur when an employee gets too close to a permanently mounted speaker inside the restaurant, or stands near a solid wall, allowing sound power to conduct from the headset speaker back to the headset microphone. Like the noise reduction function, the echo suppression DSP algorithm detects the squealing "feedback" sound within the microphone output and when detected, works to quickly change the frequency response of the amplifier to squelch the squealing sound power present in the speaker output[15].

Timeline of the Case

In 2012, the buyer engaged the designer's services to develop and design a new drive-thru system for use in the quick service restaurant industry. The top product requirement was noise reduction for both inbound and outbound paths. The second highest product requirement was that the audio quality of the system must surpass or equal the audio quality of the market-leading competitor's system. The agreed upon scope of work (SOW) included $385,000 to $475,000 cost estimate and a seven-month timeline ending in October 2013.

The design progression included obtaining and analyzing an exemplar drive-thru system from the competitor to determine how to improve upon that system's noise reduction and overall sound quality. The designer represented to the buyer that they did have the ability to meet or exceed the competitor's audio quality. The designer included this representation in their proposal of work to the buyer.

After multiple cost increases and delays, on February 10, 2016, the buyer demanded rescission of the contract. Aside from the delay and cost overrun, the major complaints voiced by the buyer were:

- Unclear and non-crisp inbound audio.
- Hum emanating through the outside speaker.
- No full duplex audio.
- Inadequate noise reduction.

In 2018, counsel for the buyer retained the author to investigate and opine on:

- Measured audio quality of the system. Did it meet specifications?
- Did the designer follow best design practices?

Author's Investigation of Audio Quality

The author visited the buyer's premises, and made subjective and objective tests, comparing the buyer's partially designed system to the exemplar system of the market-leading competitor.

Subjective Tests

Subjective tests of the partially designed system and the competitor drive-thru systems helped the author direct further objective testing. Listening to the sound quality of both systems from both the customer's (external post) and the employee's (headset) perspective, the author was able decide which audio tests would demonstrate objective, measurable differences in system performance.

For subjective tests, drive-thru posts were positioned on the back steps of the buyer's facility facing their parking lot, emulating the order-posts customers would interact with. Cables from posts were routed through the building to a base station room in the far opposite corner of the building, providing acoustic isolation.

The base station room included the buyer and the competitor base stations mounted on a wall, the buyer and the competitor wireless headsets on shelves, and test instruments on a laboratory bench.

Order Post (Outbound Path) Subjective Tests

The author first took the customer's point of view by standing outside the building and facing the order posts to interact with either the buyer or the competitor system and subjectively evaluate their outbound performance.

The buyer's representative was in the base station room, where he played the role of employee. He wore the first one, then the other system headset, without identifying which.

The test procedure for each system was as follows:

1. Half duplex evaluation

2. Full duplex evaluation

The evaluation criteria were as follows:

1. In the half duplex demonstration, one person speaks. The other listens for clarity and speech intelligibility, expecting the lack of any sounds inserted by the system that could muddle the sound.

2. In the full duplex demonstration, both persons speak over each other. The author listens for the buyer representative's voice to continue smoothly as the author and the buyer speak, without the system noticeably cutting back and forth.

Observations

1. [Half duplex] The partially designed system sounded muddled. Words were intelligible, but there were unnatural system artifacts (unexpected noise) inserted into the sound. The competitor's system sounded clear, and the buyer's words (deliberately nonsensical) were clear and intelligible in our half duplex conversation.

2. [Full duplex] With the partially designed system, chopping between our conversations was audible. The discontinuity was disconcerting. The competitor's system demonstrated proper full duplex.

Outbound Path Clarity and Intelligibility Evaluation

1. The partially designed system exhibited a constant hum. There was also clipping (system overload) when there was significant low-frequency content in the employee's speech. Clipping (resulting in harmonic distortion) and hum can be objectively measured using various instruments and measurement techniques.

2. With the competitor's system, the author could not subjectively identify any faults. To compare and confirm, the system was objectively tested with the same instruments and measurement techniques as the partially designed system.

Outbound Path Full Duplex Evaluation

1. The partially designed system full-duplex operation was flawed. This is a go/no-go test, which does not require instrumentation to prove its results.

2. The competitor's system full-duplex operation was established.

Headset (Inbound Path) Subjective Tests

The author took the employee's point of view by swapping positions. In this test, the buyer played the role of customer, and the author played the role of employee, wearing first one, then the other system headset. The buyer and the author followed the same procedure as in the order-post subjective tests and used the same evaluation criteria.

Observations

1. [Half duplex] For both the partially designed system and the competitor's systems, from the employee's perspective, words were intelligible, and there were no unnatural system artifacts present.

2. [Full duplex] With the partially designed system, chopping between our conversations was audible. The discontinuity was disconcerting. The competitor's system demonstrated proper full duplex.

Inbound Path Clarity and Intelligibility Evaluation

From the employee's perspective, the author could not subjectively identify any clarity and intelligibility faults with either the partially designed system or the competitor's system.

Inbound Path Full Duplex Evaluation

1. The partially designed system full-duplex operation was flawed. This is a go/no-go test, which does not require instrumentation to prove its results.

2. The competitor's system full-duplex operation was established.

Objective Tests of Outbound Path

The author devised objective tests confirming that low-frequency distortion was the root cause of the audible clipping and overall unpleasant sound at the customer post using the partially designed system, and to uncover any other measurable faults in either system.

Specialized audio test equipment was rented from NTI Audio[16].

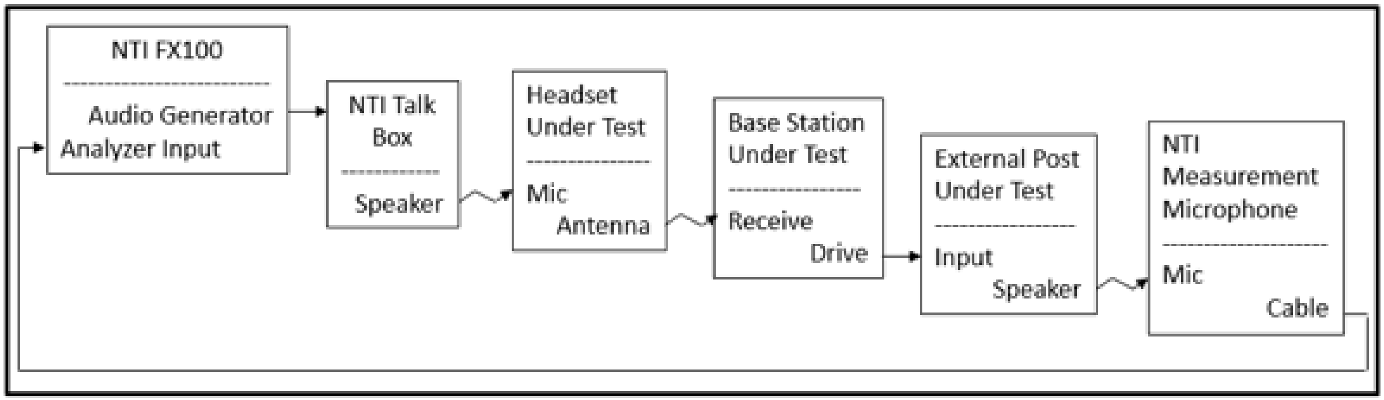

Figure 7
Test setup for outbound path.

- NTI Talk Box Speaker.
- NTI Measurement Microphones
- NTI Audio Analyzer FX100, using NTI computer software

The Talk Box can be set to present through its speaker either self-generated random audio or audio generated by the FX100 analyzer. Calibrated measurement microphones are of adequate quality to not significantly affect testing. The FX100 includes functions for signal generation, data acquisition, signal digitization, and DSP analysis. FX100 software accomplishes the control and communication of audio tests, calculation of analytical data from analyses, and generating graphical and tabular output of test results.

Tests with the NTI FX100 Analyzer confirmed objectively that the partially designed system outbound path has significant harmonic distortion in the low-frequency range. This is the root cause of the subjectively observed unpleasant audio quality from the customer post speaker.

Figure 7 shows the test setup for the outbound path. The audio generator in the NTI FX100 outputs a test signal by wire to the NTI Talk Box. The headset under test is mounted with its microphone about five centimeters in front of the speaker. The base station under test receives the wireless transmission from the headset and drives a cable to the external post under test. The NTI Measurement Microphone is positioned about one meter from the external post's speaker. The microphone cable is routed back to the NTI FX100 Analyzer Input, completing the test loop. The author performed the following tests for both the partially designed system's and competitor's headset, base station, and external post.

Glide-Sweep Tests with Audio Analyzer

For this test, the FX100 generator applies a 100 Hz sinusoid to the TalkBox for two seconds (to allow the System to settle), and then sweeps the frequency from 100 Hz to 20 kHz. The signal follows the loop indicated in **Figure 7** back to the FX100 analyzer, which records the measurement microphone's signal and performs several analyses of the recorded signal.

One such analysis is frequency response, as shown in **Figure 8**. The vertical axis of this graph is the relative sound pressure level (SPL) of the post speaker output in dBSPL. SPL is roughly equivalent to the loudness of the sound. The horizontal axis is frequency in Hertz (Hz).

The pink trace on this graph is the frequency response of the competitor's system. It shows a gradual increase from 50 dBSPL at 100 Hz to 80 dBSPL at 500 Hz. The SPL varies somewhat between 500 Hz and 3 kHz where it drops off steeply to about 30 dBSPL at about 4 kHz and remains

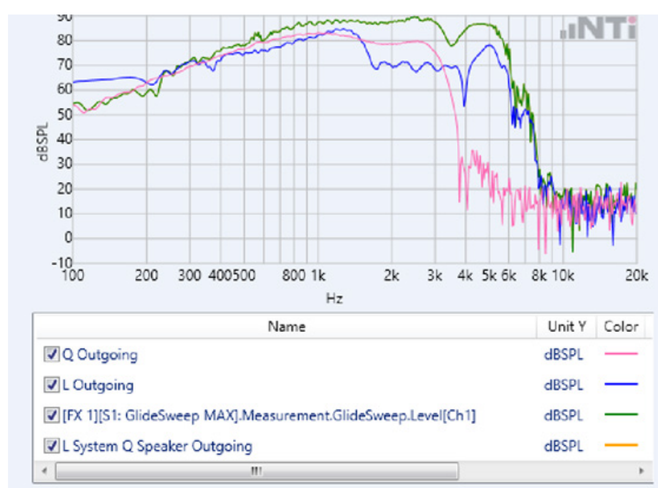

Figure 8
Frequency response test results.

substantially below 30 dBSPL from there on. One would say that the competitor's system has a 3 kHz bandwidth.

The blue and green traces are the partially designed system's frequency response. Results of two tests were saved where the green trace used a slightly higher input volume than the blue trace from the FX100 analyzer into the system. There are two apparent differences from the competitor system's performance. The SPL of the partially designed system does not drop off steeply until about 6 kHz, and more so at about 7 kHz. One would say the partially designed system has a 7 kHz bandwidth. For Hi-Fi audio, wider bandwidth is considered an advantage. But in the partially designed system, there is a wider variation in SPL between 500 Hz and 7 kHz. This approximately 30 dBSPL variation is perceptible to humans with typical hearing.

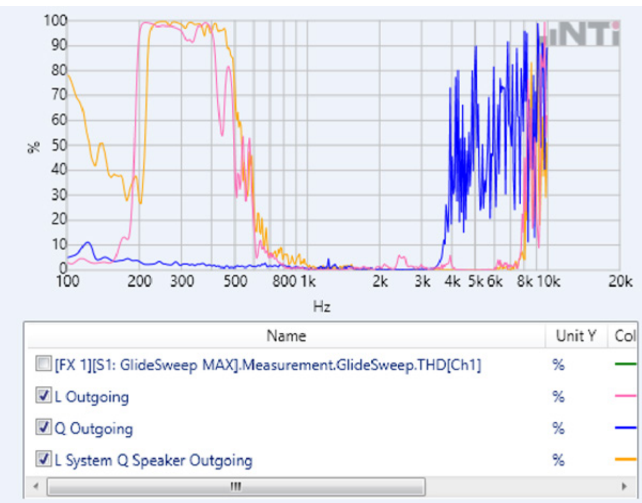

Figure 9
Distortion versus frequency test results.

The subjective observation of low-frequency distortion is objectively confirmed by measurement in **Figure 9**. It shows distortion versus frequency as calculated from the recorded glide-sweep tests. The vertical axis is total harmonic distortion in percent (%). The horizontal axis is frequency in Hz.

The blue trace is the total harmonic distortion (THD) in % for the competitor's system. It shows THD substantially less than 10% throughout its 3 kHz bandwidth. High distortion at frequencies above the bandwidth (above 3 kHz for the blue trace) is not audible. The pink trace is the THD for the partially designed system. THD is greater than 90% between about 200 and 400 Hz and does not fall to less than 10% until about 700 Hz. This shows objectively that there is significant, measurable distortion at low frequencies. The orange trace corresponds to the retest of the partially designed system using the competitor's external post, ruling out the external speaker posts as the cause of distortion.

Single-Tone Tests with Audio Analyzer

The author ran single-tone distortion tests at several frequencies to examine THD more closely in the partially designed system.

As a reference point, **Figure 10** shows the spectrum of the recorded output of the partially designed system's

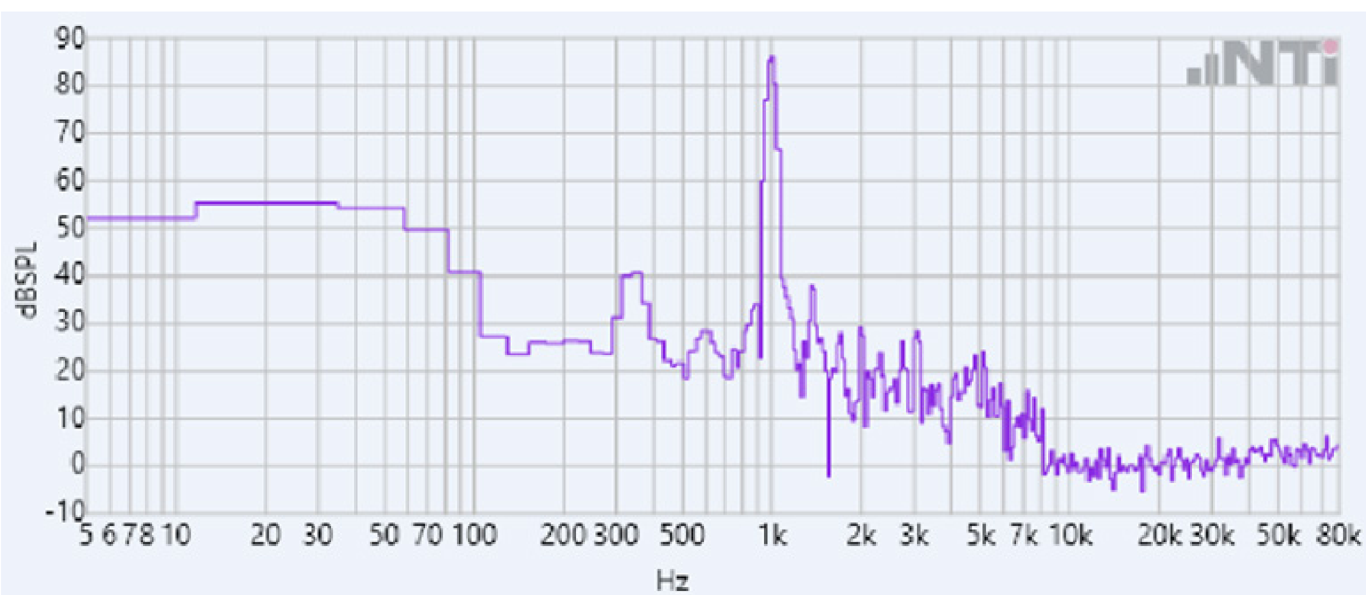

Figure 10
The partially designed system buyer's system output spectrum with 1 kHz input.

external post speaker when a 1 kHz tone is applied by the NTI Analyzer. The vertical axis is in dBSPL, and the horizontal axis is in Hz. The magenta trace is the relative loudness at each frequency. At 1 kHz, the magnitude is about 85 dBSPL. If there were significant harmonic distortion, we would see similar spikes at 2k, 3k, and higher multiples of 1 kHz. At 1 kHz, there is no significant harmonic distortion. However, notice the raised area of the trace between about 300 and 400 Hz, which reaches about 40 dBSPL. This range of frequencies corresponds to the audible hum of the partially designed system.

A 500 Hz input tone caused the worst measured distortion, as shown in **Figure 11**. At 500 Hz, the magnitude is a little more than 80 dBSPL. At that frequency, at one or more points within the partially designed system, the 500 Hz magnitude is so large that the system is overloaded. Instead of a smooth sinusoid, the signal is "clipped" so that it is flat at the top and steep at the sides.

The spectrum shown in **Figure 11** is consistent with a clipped sinusoid. In addition to the spike at 500 Hz, there are spikes at 1k, 1.5k, 2.5, etc. A signal with this spectrum will sound clipped to typical human perception, and speech through such a system will not sound as clear and crisp as speech through a system without such distortion.

Notice the **Figure 11** spectrum also shows the system-

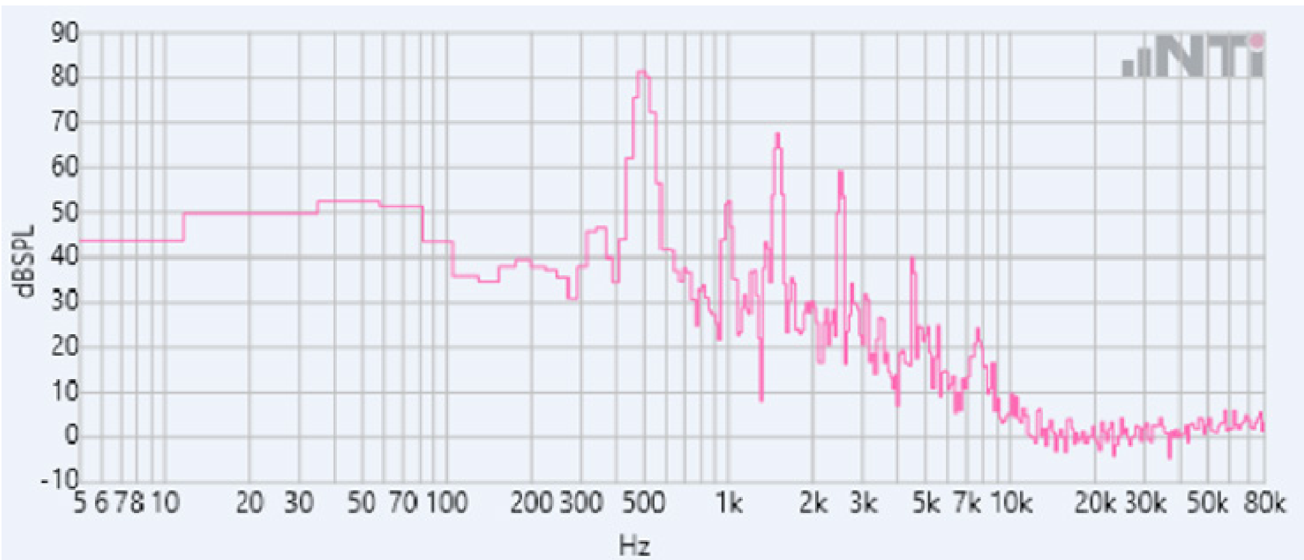

Figure 11
Partially designed system buyer's system output spectrum with 500 Hz input.

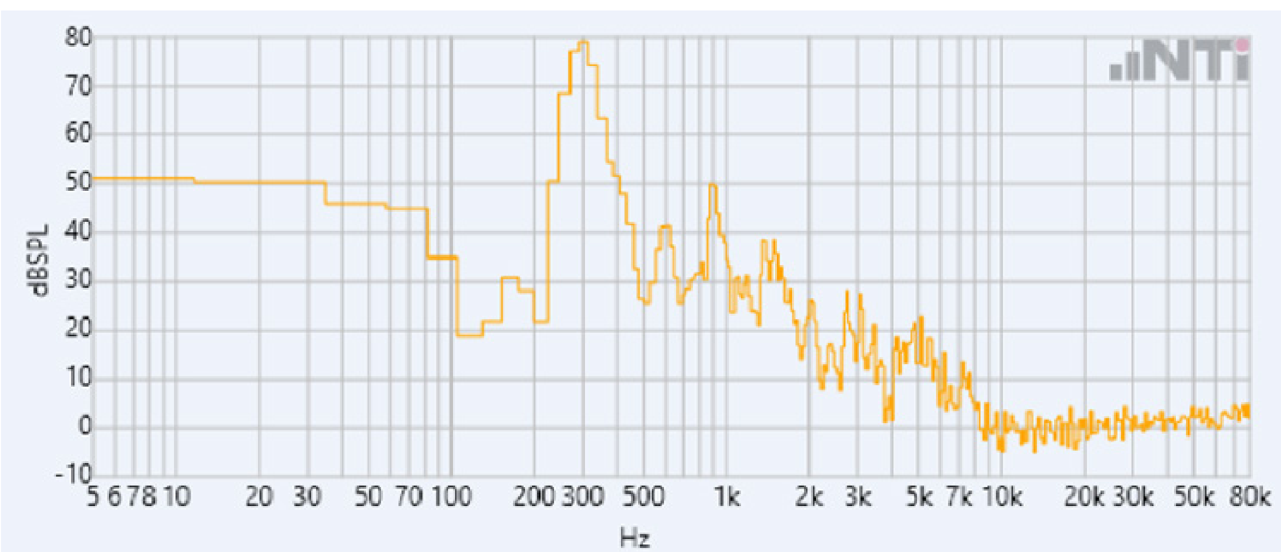

Figure 12
Partially designed system buyer's system output spectrum with 300 Hz input.

generated hum between 300 and 400 Hz. To explore that 300 Hz hum more closely, the final test applied a 300 Hz sinusoid with results shown in **Figure 12**.

The expected spike at 300 Hz is there, at about 80 dB-SPL. Distortion tones are visible at 600 and 900 Hz. Notably, the 300 Hz spike is widened by the system-generated hum.

Conclusions Regarding Audio Quality

The communication system that the designer delivered to the buyer did not meet the agreed specifications of the system requirements documents or the contract. This was shown by objective measurement/testing and confirmed by subjective listening tests.

The measured low-frequency distortion and hum of the buyer's outbound system are much worse than the same measurements for the competitor's system. The higher bandwidth of the partially designed system is not an advantage.

Best Design Practices

The author had been asked to opine on whether the designer followed best design practices.

Linear Time Invariance Non-Linear Time-Varying Systems

The audio system of a drive-thru lane is a non-linear, time-varying system. Its complexity arises from competing functions, including automatic gain control, noise reduction, echo suppression, and full-duplex operation, which involve multiple interdependent feedback control loops.

In contrast, feedback loops in linear, time invariant (LTI) systems are independent. LTI systems are easier to design than non-LTI systems using hand or spreadsheet calculations and engineering reasoning. See references for a formal mathematical article on LTI systems[17,18]. A good article about nonlinear time-varying systems can be found here[19].

To explain linearity, let' s say the input from a microphone to a linear amplifier is "x", the amplifier gain is "G", and the amplifier output to the speaker or headset is "y". Then if the amplifier is linear, "y = G*x". If the input becomes twice as loud, 2x, then the changed output (call it "z") is also twice as loud "z = G*2x", so "z = 2*y". When you wiggle the input x, the output wiggles the same way, but G times bigger.

G may be different for different frequencies in an LTI system. The relative size of G at different frequencies is called its "frequency response." In a typical audio system such as the drive-thru system, G is very small at frequencies higher than the system's bandwidth.

Time invariance means that in the equation "y = G*x", it doesn't matter when "x" is applied to the input. G remains constant over time, even if G is different for different frequencies of "x". To put it mathematically, if the value of x at time t is x(t), then y(t) = G*x(t). In a time-invariant system, for any time T, y(T) = x(T). In a time-varying system, G changes with time. For example, when a soft-spoken customer is at the post, G increases from its level for a loud customer.

An audio system with a maximum output loudness is an example of non-linearity. Let's say that its maximum output loudness is K. While "G*x" is less than K, then "y = G*x" is true, and the system is operating in its linear region. However, while "G*x" is greater than K, then "y=K". While "G*x" is greater than K, wiggling x does not change y. For this situation, the system is operating outside of its linear region and "y" is said to be clipped.

Assuming LTI and independent feedback loops, one may devise mathematical formulas for analyzing frequency response and the stability of control loops — and these formulas may serve as the basis upon which to design the system. However, the AGC, NR, full-duplex operation, and echo suppression functions that were to be present in the buyer's drive-thru system are all non-linear and time-varying functions. Designing a system with all these functions is not straightforward.

As described previously, the AGC function works to keep the speaker/headphone sound volume "y" constant. If the microphone output "x" is small, the AGC increases "G". If "x" is large, the AGC decreases "G". Thus, "G" is not constant over time, and the system is "time-varying."

The noise reduction function of the processing algorithm detects noise sound within the microphone signal "x." When interference is detected, the algorithm works to quickly change the frequency response of the system to decrease the noise power present in the speaker output "y" In one sense of echo suppression, the DSP echo suppression function works in a similar manner, detecting the squealing echo pattern and squelching it. In both noise reduction and echo suppression (both squelching and double-talk) functions, the system is time varying.

The full-duplex operation of the system is also non-linear and time-varying. In this duplex system, both the customer's and the employee's microphones and speakers are always on. Duplex operation requires a non-linear control loop algorithm to create a natural sounding conversation such as in telephone conversations.

Because these four control loops are interdependent, they get in each other's way. For one example, a large interference sound from the microphone "x" may incorrectly cause the AGC loop to decrease the system gain "G" to keep speaker output "y" constant. This was indeed the case, as communicated by the buyer in July 2015: "As the background noise gets louder, the inbound microphone volume decreases."

Designing a non-linear, time-varying system is more difficult than designing a linear, time-invariant system. Modern design methodologies for non-linear, time-varying systems use model-based design and functional simulation early in the design process to help the designer understand and anticipate design challenges before designing or buying hardware or writing or buying software code. Model-based design entails simulating the functionality of a system comprised of behavioral models. Functional simulation, behavioral simulation, and model-based simulation are synonymous.

Functional simulation entails the creation or purchase of functional behavioral models for each subsystem. The models mathematically describe how each subsystem's outputs change over time as its inputs change over time. Models are written to reflect the subsystem's non-linearities and time dependencies. Subsystems represented by models are interconnected to form the top-level system. A simulation testbench interconnects the top-level system with models that provide stimulus signals to system inputs and models that examine and analyze system outputs. Tests are written to exercise the system inputs in ways that verify the system reacts properly to the inputs.

Model-based design and simulation of complex electronic systems has been the design flow best-practice since the mid-1990s. Based on the documents and emails provided, the designer did not use a model-based design methodology.

The following is a partial list of electronic design automation (EDA) tools capable of model-based design that were available to the designers before 2012. Since then, some of the vendors have consolidated, and new vendors have joined the market. The list is not intended to be all-inclusive, only to show that there were multiple options for incorporating model-based design:

- The Mentor Graphics "PADS" EDA platform[20] (which was available to the designer). The PADS EDA platform is capable of model-based design using its VHDL-AMS simulator. VHDL-AMS behavioral models could have been purchased from component vendors or written in-house for circuitry and algorithms that were to be designed by the designer.
- A similar EDA platform sold by Cadence Design Systems[21] uses either Verilog-AMS or VHDL-AMS.
- The Advanced Design System (ADS) sold by Keysight (formerly Agilent Systems, formerly Hewlett Packard) is an EDA platform for RF, analog, and digital system design, which includes signal generator models which correspond directly to Keysight's laboratory instruments[22].
- Several companies produce prototyping systems that incorporate MATLAB with Simulink and HDL-Coder to drive electrical signals into hardware and receive electrical systems from hardware. This is known as "Hardware in the Loop" verification[23].

The drive-thru system is a non-linear, time-varying system that was to include at least four interdependent control loops of automatic gain control, noise reduction, echo suppression, and full-duplex operation. Integrating these competing functions was complex because a change to one component may cause its function to interfere with the function of another component.

Best practice is to design and verify the noise reduction, automatic gain control, and echo suppression functions in tandem. The designer's assumption that third-party algorithms can be plugged into the overall DSP program was simplistic. A reasonable approach is to use models of third-party components in the system simulator of choice, discover where they interfere with each other, and modify the software components accordingly.

Given the difficulties of designing a complex, non-linear, time-varying, multi-feedback-loop system such as the drive-thru system, it was an unreasonable decision by the designer to attempt the project without taking advantage of best practices for modern design methodology. The designer had a choice of EDA platforms, modeling styles, or simulation engines to use and present to the buyer; choosing "no system simulation" was an unreasonable choice that deviated from best practices in the field, was a major contributor to the project being late, over budget, and incomplete, and ultimately resulted in the designer's inability to provide a system with the required feedback loops after working on the project for nearly three years.

Conclusions Regarding Design Practice

a. Solving the complexity of the required system was beyond the designers' capability using Linear Time-Invariant design techniques. The limitation of Linear Time Invariant design approach is taught at the undergraduate level to electrical engineering majors.

b. The designers should have realized upon reviewing the requirements document that the well-known difficulty of stabilizing a control system comprising multiple non-linear, time-varying feedback loops without the use of system simulation software programs was impractical using linear time-invariant design methodology.

c. When outsourcing software design of individual system components, the designers should have provided a top-level framework for simulating and verifying the software with hardware.

Resolution

The author submitted an expert report late in 2018. The designer and the buyer reached an agreement soon thereafter. The author was not privy to the terms of the agreement, but presumably the expert report was convincing to both parties.

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