



Using an Adaptive Voice User Interface
to Improve Customer Service and
Reduce Operational Costs

An Interactive Digital White Paper

www.interactive-digital.com

Technology Background

While user- personalization has had demonstrated success for Web based interactions, it has yet to be fully leveraged in the handling of Automated Calls.

Advances in speech technology such as ASR, Natural Language Understanding, Caller-Directed Dialogues and Web Profiles are excellent enabling technologies, though these are only part of the solution. Used in conjunction with a well designed Voice User Interface or VUI, they represent a significant improvement over earlier technologies. However, one important design factor that has until recently been overlooked, is the individuality and in-call behavior of the telephone callers themselves.

Every caller to your speech application has his or her own individual set of aural, speech, hand-eye coordination (as used in DTMF keypad entry) and material comprehension skills. Add to this environmental variables such as background noise, poor mobile phone signals and caller distraction and it becomes clear that each call to the IVR System is truly a unique interaction.

This is one of the main reasons human operators are so good at handling any type of call – they can handle the dynamics of human conversation intuitively and with ease. Your callers know this all too well and will opt for an agent the first time your speech system fails to be productive for them.

To the extent that the speech application can monitor and adjust to the behavior of a particular caller during the call, a proportionate number of your automated calls can be more efficient and productive. While a well designed call script with optimal structure and content, intentional pauses, grammar tuning and context forming are excellent design principles, the system falls short if it does not consider the real-time behavior of the caller just as a human would under the same circumstances.

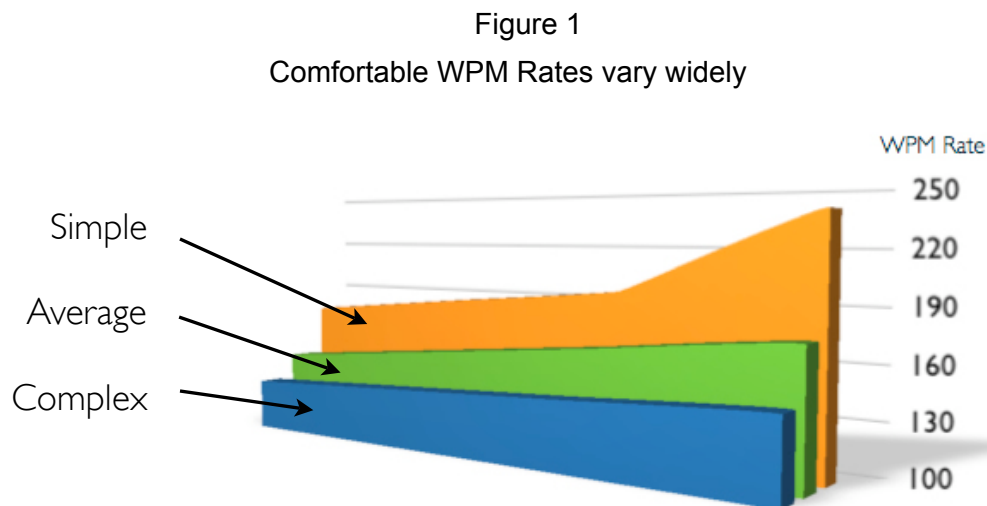
This white paper describes how the use of a real-time Adaptive VUI can be used in conjunction with best practice design principals to provide optimal use and efficiency gains for your automated speech applications.

Auditory processing is an individual thing

Human conversation is a dynamic and highly individualized process

Research shows that the average English-speaking rate is 130 - 200 Words Per Minute. This wide WPM range applies to 90% of the English-speaking population.

- For complex material, a rate of 130 - 145 WPM may be required
- For material of average complexity, 145 - 175 WPM can be optimal
- For simple material, many listeners can accommodate over 175 WPM



Listeners can be lost to boredom, lost to complexity or fully engaged in a conversation based on the speaker's ability to deliver all types of material at the optimal rate for each listener.

Good communicators are aware of this and continuously monitor their audience. They periodically adjust their conversational pace, verbal content and emphasis to get the message across effectively and efficiently.

They make these adjustments in an instinctive, fluid and natural way, thereby quickly “tuning in” to establish optimal harmony with the listener and keeping them fully engaged in the dialogue. In summary:

- An effective speaker must first be a good listener.
- Adjusting the conversation to suit the material and the abilities of the listener in real-time is essential to becoming an effective communicator.
- Automated Call costs can be reduced and the caller experience can be improved to the extent IVR systems can accomplish this natural and instinctive human characteristic.

Compromising individuality has consequences

Disregard the listener and you risk disengagement

Traditional speech applications are “static” and make no dynamic adjustments for the real-time behavior of your individual callers. As a result, all callers are handled in the same way regardless of their knowledge, experience, navigation skills and willingness to use your automated system.

Specifically, all audio content is delivered to the caller at the same WPM rate regardless of their demonstrated behavior during the call. These applications do not listen for signs that the listener (your customer) understands what is being said and is comfortable with the pace and content of the dialogue. Without “tuning in” to a callers behavior during the call, real IT efficiencies are lost.

As shown in Figure 2, this can lead to consequences in terms of customer satisfaction and the costs associated with the handling of automated calls.

Figure 2
Human Factors - Compromising has its Consequences

Human Factor	Compromise	Consequence
Caller Skills: Individual Conversational and Speech/Keypad navigation skills.	Automated Calls are not tuned to each individual callers skill levels.	Call is longer, less productive and less efficient than it should be.
Caller Attitude: Level of like or dislike for Automated Voice Systems in general.	All Callers handled in the same way regardless of acceptance for automated voice systems.	Marginal users are more inclined to opt for an Agent.
Environment: Callers may be using a home, mobile, car, public or office phone.	No real time adjustment for real world dynamics.	Even accepting and power users can get frustrated.
Attention: Callers (expert and novice alike) can easily be distracted during a call.	No allowances for the callers circumstances.	Caller knows an agent will understand.
Familiarity: All callers are not equally familiar with the voice application. This is becoming an issue with the trend towards automating increased amounts of information.	No provisions for less familiar/unfamiliar callers.	Caller knows they can ask the agent anything.

When your callers can interact with a system that is tailored to their specific set of skill levels, knowledge and aural, vocal and hand-eye coordination talents, many efficiencies are gained in the handling of your automated calls including:

- Highly skilled power users can more quickly navigate the application flow
- Greater tolerance for callers having difficulty hearing/understanding the audio
- Unfamiliar callers are given a reason to become repeat users
- Salvaging the automated call for callers using mobile or public phones.
- Distracted callers get the flexibility they need

Approximating the human response

Caller adaptive interactions increase IVR effectiveness

From the preceding, it is clear that having your automated speech system listen to how your callers behave during the call and adjusting the audio responses of the system accordingly promotes improved communication. Every individual caller has their own natural conversational rhythm; a dialogue pace at which they are most comfortable and productive. This form of “tuning in” to the individual caller during the call emulates the principals of good human-to-human communication. The Adaptive Audio software product from Interactive Digital accomplishes this for the majority of automated speech systems available on the market today.

The product is available for the developer for use in both open standards like VoiceXML and for proprietary IVR solutions. Changes to the speech system hardware, operating system and/or application development tools are NOT required to implement Adaptive Audio and production pilots can be up and running in a very short time.

When added to your speech applications, the product allows your automated IVR System to continuously monitor individual caller behavior during each call to the system. Speech and/or DTMF responses are monitored for speed and accuracy in real-time, node-by-node in the call script. The product measures for example, how long each individual caller takes to enter account numbers, a PIN, their telephone number and any other application specific data required by your application. Additionally, how long callers take to select from menus or speak their desired choices are measured for both speed of entry and accuracy.

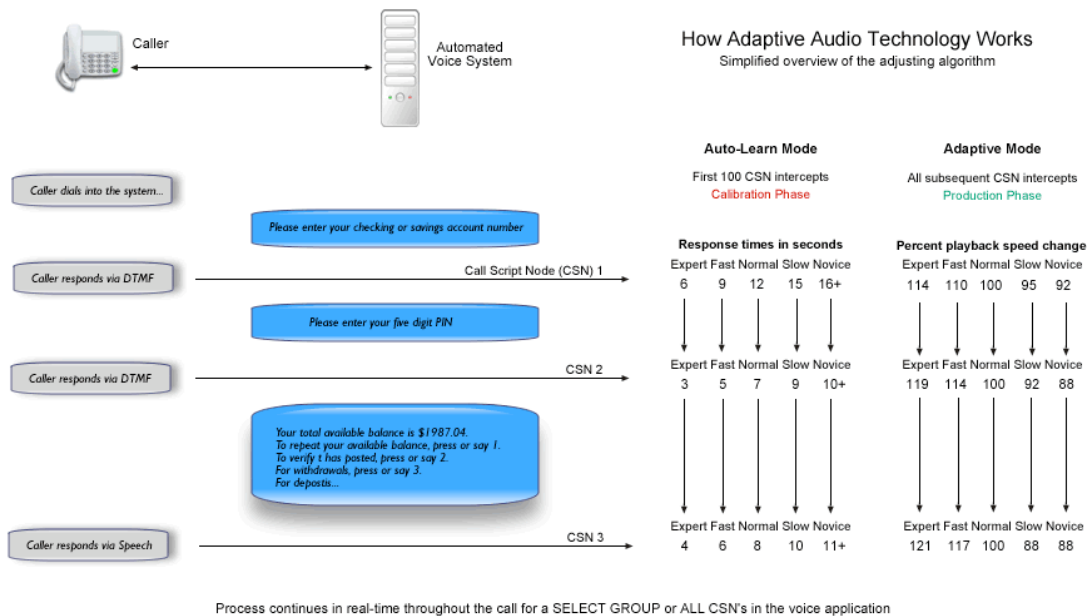
When run for the first time, the software listens and learns how effectively your callers navigate each of these script nodes in your voice application. No speed or message content adjustments are made for the first several hundred passes through each node. It is during this auto-learn stage that the product gathers the caller and application specific information it will need later on to make intelligent

decisions as to when and how to adjust the WPM and/or content for a caller of a particular skill level.

Once the product has acquired a sufficient calibration sample, it automatically switches to adaptive mode. Here the software uses the previously stored behavioral information to automatically adjust the WPM speaking rate and voice message content of the system to suit the skills and exhibited behavior of each individual caller in real-time. This personalizes the call experience as it happens, creating a friendlier, more responsive and productive customer experience.

Figure 3 below illustrates how the basic speed adjustment part of the algorithm works.

Figure 3
Dynamic adaptation and personalization throughout the call



Adaptive Audio is Patented. U.S. Patent No. 5,493,608. Other Patents Pending.

Using Adaptive Audio, the system administrator can set minimum, intermediate, and maximum playback speeds for your speech system, as well as the caller response times that will trigger a change in playback speed or message content.

In addition to the experienced caller, this solution also detects slower response times and can slow down the IVR playback to allow an inexperienced caller extra time to hear and understand the available options and prompts.

You can target Average Handle Times, Average Handle Rates or a combination of these as benefits the product delivers. Forced playback speed presets and timing delays are also supported for fine-tuning the application for optimal performance.

This process works autonomously for every caller without the need for ANI, customer databases, and web based profiles. Prior knowledge about your callers is not required and the process works every time in real-time. This is important because even power users do not behave the same way when distracted or on a cell phone and ANI based profiles are not helpful when a caller is using a different phone to call your application.

The product provides statistical reports that indicate how long callers are taking to respond to each node in the application, thus allowing you to identify potential design problems in the call script.

Software component overview

Flexible design for easy integration on any platform

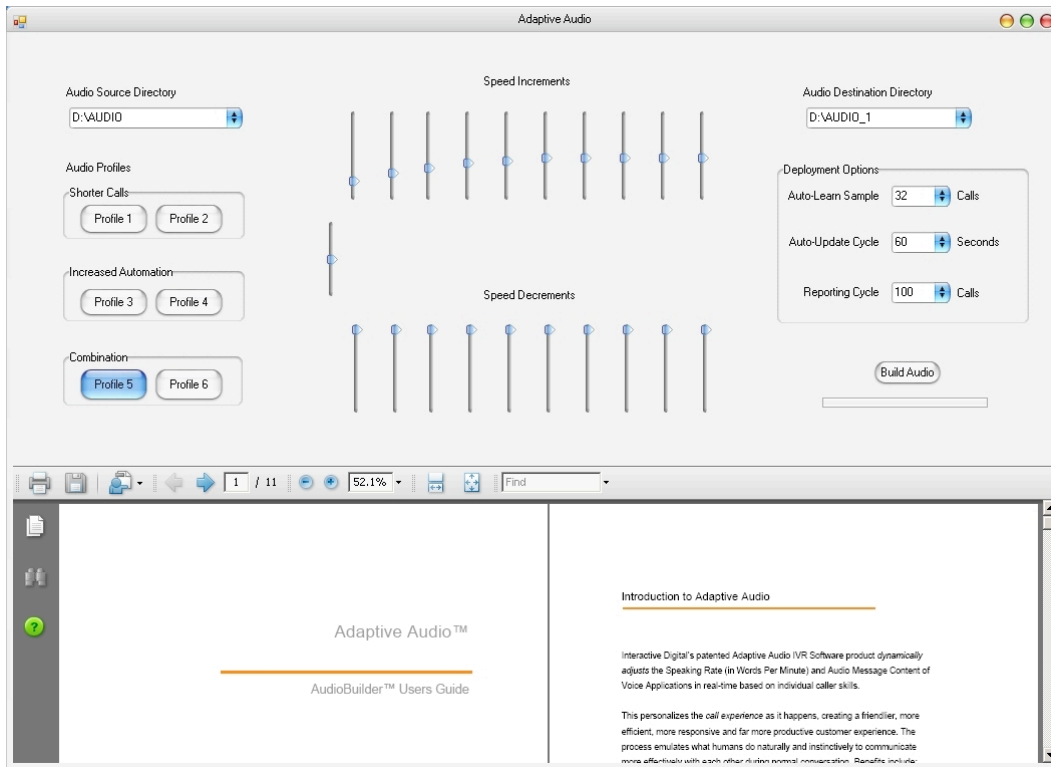
Adaptive Audio software consists of two principal components:

AudioBuilder™ - This is a standalone Windows based tool that builds and maintains the alternate speed audio files needed at run time for Adaptive Audio. Figure 4 shows the user interface for this software.

AudioBuilder is used when the IVR platform in use does not support dynamic playback control for pre-recorded audio files. The tool builds and automatically

maintains the required audio as files change over time and runs on the applications remote Audio Server or premise based IVR.

Figure 4
The AudioBuilder User Interface



AdaptiveAPI™ - This is the Adaptive Audio software component that automatically tracks caller responses and selects appropriate voice playback speed and/or content. This component is implemented as a discrete, easily integrated platform certified library module that is called throughout the voice application. For Unix based platforms, this would come as a shared object file. For Windows, it would be a Dynamically Linked Library or DLL.

The Core callable functions of the AdaptiveAPI are:

- adaptiveAudioStart()
- adaptiveAudioAsk()
- adaptiveAudioAnswer()

It is necessary to place a single call to the `adaptiveAudioStart()` function at the start of each call. This function initializes the per call session data and must be placed immediately after the application answers an incoming telephone call.

Thereafter, successive calls to the `adaptiveAudioAsk()` and `adaptiveAudioAnswer()` functions are placed at the beginning and end of each point at which the voice application prompts the caller for input.

Once implemented for a specific voice application, this process does not have to be repeated. Slight modifications may need to be made, but only if the application changes significantly.

Additional capabilities with Adaptive Technology

Adaptive interactions go beyond audio speed and content control

Besides adjusting audio playback rates and message content, the following features are implemented within the Adaptive Audio product.

1. Best Modality Signaling (BMS) - Informs the voice application whether Speech or DTMF input has historically been more efficient and/or more successful by a significant margin at various points in the call script. The qualifying thresholds are user programmable in the `AdaptiveAPI.cfg` file.

Feature Benefits: Increases call automation rates and reduces user error rates in the voice system.

2. Adaptive Timeout Control (ATC) - Allows the voice application to dynamically extend timeout periods for individual callers having significant difficulty navigating areas of the call script. Since Adaptive Audio is aware of when each individual caller is experiencing difficulty navigating any or all of the call script, it can inform the voice application as how much additional time (in one second intervals) should be added to an existing timeout value to allow such a caller to respond.

Feature Benefits: Increases automation rates and reduces error rates and unwarranted CSR transfers.

3. Preemptive Transfer Alerts (PTA) - This feature keeps a cumulative index of how well each individual caller is navigating the call script and Identifies callers having excessive difficulty navigating the voice application. When such callers are identified by the Adaptive Audio software, it recommends preemptively transferring them to a CSR. Thresholds are user programmable and PTA signals factor in the likelihood that a CSR is available based on incoming call volume.

Feature Benefits: Reduces call backs and abandoned calls and increases customer satisfaction with the voice system.

4. Dynamic Application Smoothing (DAS) - Dynamically adjusts the WPM speaking rate up or down for any points in the call script that callers find particularly easy or difficult to navigate. Adjustment decisions are based on the level of difficulty of each CSN¹ as represented by the historical behavioral data collected at each CSN by Adaptive Audio. This feature is fully automatic once optioned in the Adaptive Audio configuration file and no further action is required by the developer in order for the process to take place.

Feature Benefits: Shortens average call duration, reduces error rates in the voice system and provides an improved user experience.

5. Application Dependent Profiles - Provides independent control over the audio playback rates, audio content and adaptive patterns used in multi-application environments. A requirement for proper functioning of this feature is that there is no overlap in CSN use between applications.

Feature Benefits: Shortens average call duration and reduces user error rates.

6. Caller Behavior Analytics - Provides real time, comprehensive analysis and reporting on caller behavior, expertise and willingness to use the voice application. Pinpoints application trouble spots and indicates where the application design can be improved. Also included are difficulty ratings for each CSN and Adaptive versus Non-Adaptive performance comparisons. Report data is provided via output to a plain text log file that can be viewed online or via a spreadsheet tool. Figure 5 shows a section of how the report is organized.

Feature Benefits: Provides a unified report for a better understanding of how and where improvements can be made in the voice application script.

Figure 5
 Caller Behavior Analytics Report

Adaptive Audio - Caller Behavior Analytics
 Adaptive API Software Version 5

Voice Server Report Summary

Voice Server: Adaptive-1
 Report Date: Sat Aug 09 16:47:33 2008
 Calls Answered: 40
 PTA Signals: 2

Voice Server Configuration Settings

Calibration Size: 8 Calls
 Audio File Maintenance: 10 Sec.
 Log Update Every: 8 Calls
 Non-Adaptive Control Port: 0
 Adaptive Monitor Port: 1
 BMS Sensitivity: 15 Percent
 PTA Threshold: 10 Percent
 DAS Decelerate Trim: 20 Percent
 DAS Accelerate Trim: 80 Percent
 Speed Profile 1 CSN Range: 1 - 133
 Speed Profile 2 CSN Range: 499 - 512
 Speed Profile 3 CSN Range: 719 - 1023
 Audio Playback Speeds: 92, 95, 104, 100, 107, 110, 112, 115, 118, 121

Call Script Node (CSN) Profile Report

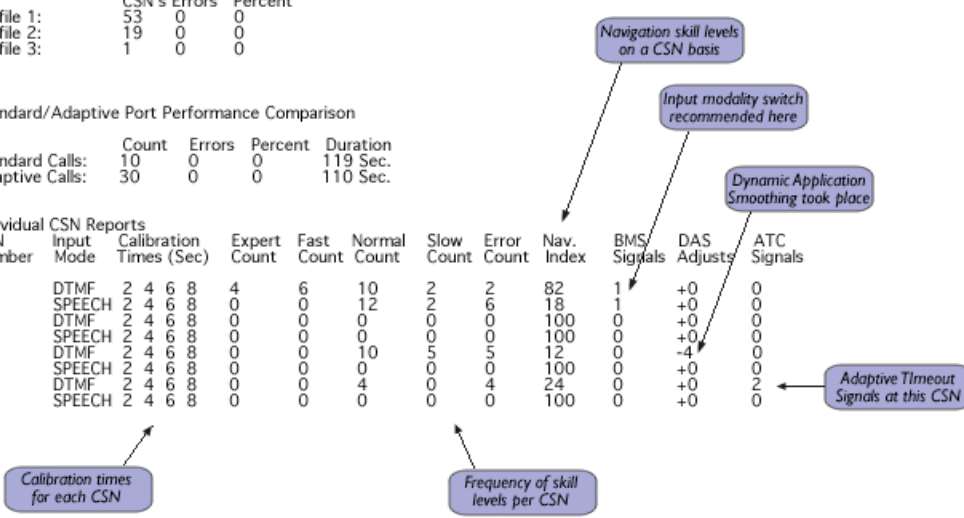
Profile	CSN's Errors	Percent
Profile 1:	53	0
Profile 2:	19	0
Profile 3:	1	0

Standard/Adaptive Port Performance Comparison

	Count	Errors	Percent	Duration
Standard Calls:	10	0	0	119 Sec.
Adaptive Calls:	30	0	0	110 Sec.

Individual CSN Reports

CSN Number	Input Mode	Calibration Times (Sec)	Expert Count	Fast Count	Normal Count	Slow Count	Error Count	Nav. Index	BMS Signals	DAS Adjusts	ATC Signals
0	DTMF	2 4 6 8	4	6	10	2	2	82	1	+0	0
0	SPEECH	2 4 6 8	0	0	12	2	6	18	1	+0	0
1	DTMF	2 4 6 8	0	0	0	0	0	100	0	+0	0
1	SPEECH	2 4 6 8	0	0	0	0	0	100	0	+0	0
2	DTMF	2 4 6 8	0	0	10	5	5	12	0	-4	0
2	SPEECH	2 4 6 8	0	0	0	0	0	100	0	+0	0
3	DTMF	2 4 6 8	0	0	4	0	4	24	0	+0	2
3	SPEECH	2 4 6 8	0	0	0	0	0	100	0	+0	0



CSN¹ - Call Script Node. This is the point at which the voice application prompts the caller for input. The start of a CSN is defined as the beginning of voice play for the first voice segment of a CSN. The end of a CSN is defined as the point at which the application receives a terminating event, either valid or invalid from the caller.

Production Metrics show operational efficiency gains

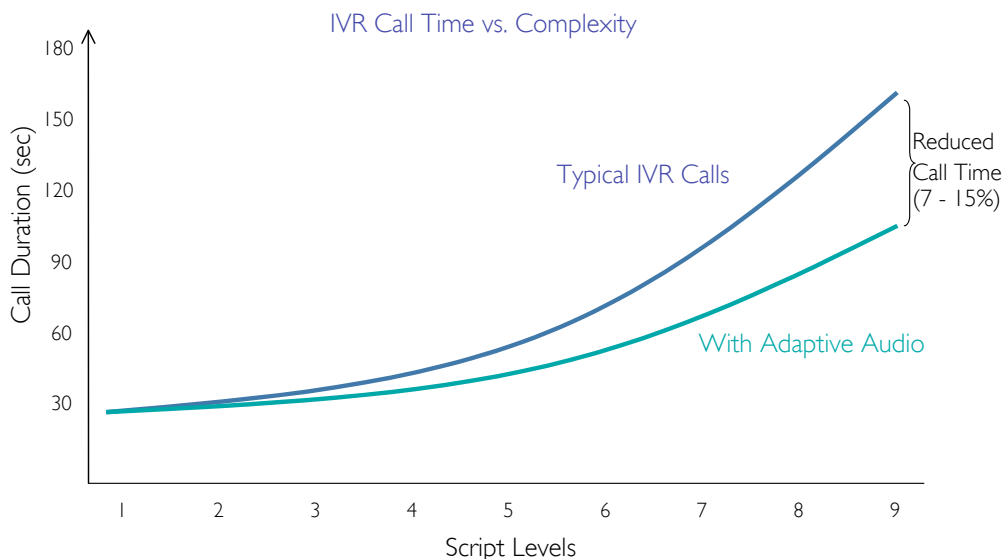
Benefits depend on the voice application

The benefits of an Adaptive VUI vary based on the design, content and average call duration of the speech application. Applications must provide a sufficient amount of caller interaction to make the technology worthwhile.

As a rough estimate, a minimum average automated call length of 40 seconds with at least three caller responses is needed to make adaptive technology worth implementing. In general, the more levels of scripting and the higher the average automated call duration, the greater the benefit.

Based on production metrics gathered at various Adaptive Audio sites, Figure 6 illustrates the relationship between automated call script length, script levels and the effectiveness of the adaptive process.

Figure 6
Adaptive calls are more efficient

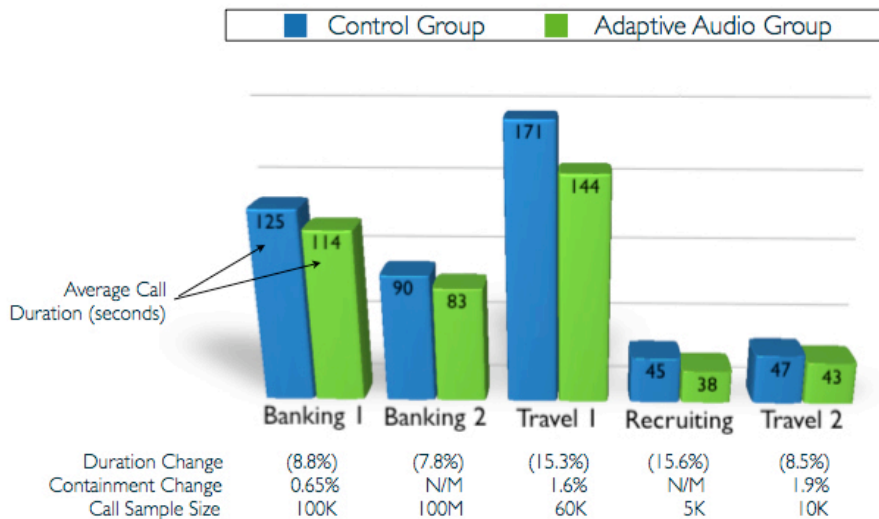


The data in Figure 7 illustrates production results gathered from various types of voice applications that leverage Adaptive Audio. This data indicates improvements in both the Average Handle Time (AHT) and Average Handle Rate (AHR) achieved with the product.

An interesting statistic here indicates that the Rail Travel application experienced a drop of 27 seconds in AHT, while the AHR simultaneously increased by 1.6 percent. This particular implementation of Adaptive Audio used only positive playback increments of 110, 114, 117, 119 and 121 percent. Normal playback here is defined as 100 percent and the usual decrements of 95, 90 and 85 were not used as the customer was targeting better AHT improvements only.

This indicates that power users are more likely to stay with the automated system provided it moves at their comfortable speaking rate – even though that rate is 10 – 21 percent faster than it would be without Adaptive Audio. In summary, this proves the earlier assertion that tuning in to the natural rhythm and pace of the caller helps keep them engaged in the automated call process.

Figure 7
Production results with Adaptive Audio



Business Case for Adaptive Audio

Measurable efficiency gains reduce operating costs

For typical B2C Retail, Financial, Travel and Government applications, savings of 1 – 5 percent in AHR and 7 – 15 percent in AHT can be expected when incorporating adaptive functionality. This translates into significant cost savings since automated speech and touch-tone calls cost about \$.75 each to answer while agent handled calls are about \$4.25 on average.

As a result of shorter calls, increased automation and an interface that improves the caller experience by adjusting to their skill levels, the following benefits are provided:

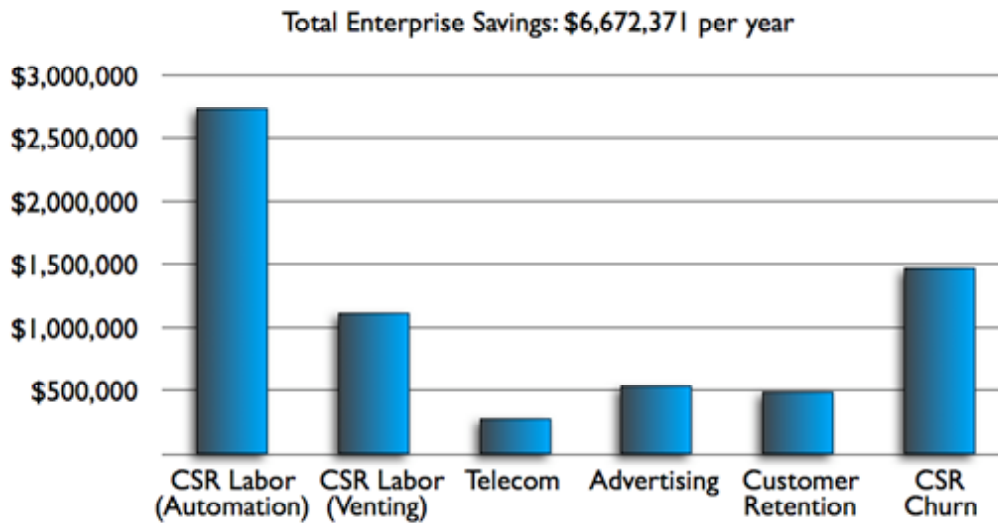
- **Reduced CSR Labor Costs Due To Increased Call Automation:** If self-service is a quicker-and-easier experience for the caller, they will be more likely to use it and stick with it once they have started to use it. Additionally, this results in an increase in customer satisfaction and faster response times.
- **Reduced CSR Labor Costs Due To Fewer Callers Venting:** The average CSR talk-time will be longer for callers that are upset and irritated with an ineffective IVR. These callers will often take time to complain, which takes the time of the CSR.
- **Reduced Telecom Costs:** Shorter automated calls are a product of a user experience that is tuned to each individual callers skill level. This results in lower telecom operating expenses for the call center.
- **Reduced CSR Churn:** Callers invariably have a strong dislike for self-service. One of the reasons for this is that the prompts are long and tedious and not able to accommodate their skill level. Adaptive Audio Technology improves the caller experience and results in fewer complaints to the CSR. This makes the CSR's job a bit more pleasant, thereby reducing CSR churn and increasing CSR Productivity.

- **Reduced Advertising Costs:** An enterprise that irritates callers by having a self-service that is disliked will need to spend more on advertising to compensate for this.
- **Increased Customer Retention:** Previous studies have shown that the cost to acquire a new customer is approximately five times the cost of keeping that customer. Satisfying customers will contribute to retaining their business and decrease the cost of acquiring new customers.

Figures 8 and 9 below illustrate a cost savings analysis performed for a hypothetical Financial Services organization. The savings calculations shown here were determined using data from production runs with Adaptive Audio at customer sites and a comprehensive spreadsheet ROI calculator developed by an independent research firm. Variables used in the calculations are as follows:

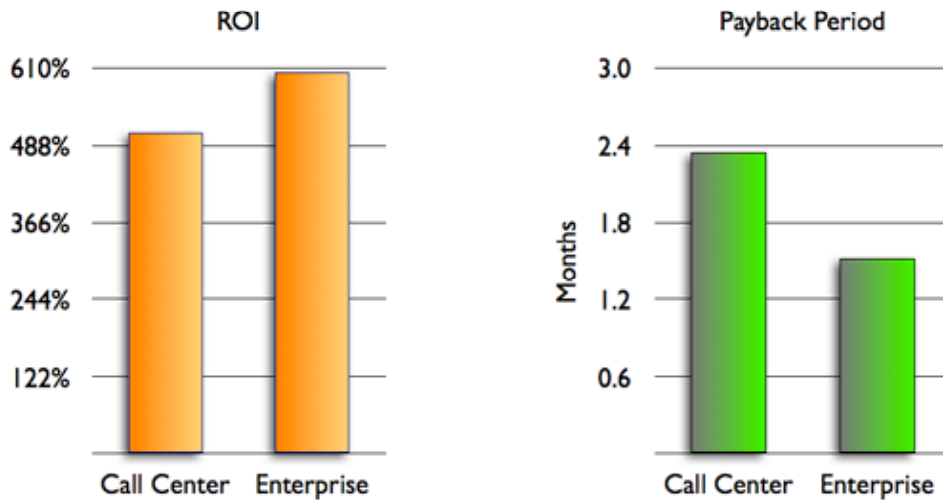
Monthly Call Volume: 10M Calls Average Call Time: 280 Seconds
 Automation Rate: 70 percent Self-Service Duration: 100 Seconds

Figure 8
 Adaptive Audio savings in Enterprise Operational Costs



Obtain the ROI Calculator via email at sales@interactive-digital.com

Figure 9
Adaptive Audio ROI and Payback Periods



Figures 10 and 11 on the following pages illustrates a cost savings analysis performed for a typical medium and large sized call centers.

Figure 10 shows the results for a call center handling 2 million calls per month with an average automated call duration of 130 seconds. Typical ROI at a site like this will be just over 456%, with an enterprise payback period of about 2 months and a total enterprise cost reduction of over \$1.34M per year.

Figure 11 shows the results for a call center handling 10 million calls per month with an average automated call duration of 110 seconds. Typical ROI at a site like this will be just over 550%, with an enterprise payback period of 1.66 months and a total enterprise cost reduction of over \$6.69M per year.

Figure 10
Adaptive Audio savings calculations for a medium-size Call Center

Call Center Variables:		
Calls per month	2,000,000	
Average length of talk time per call (seconds)	280	
% of calls that are completed by self-service	70%	
Average length of self-service call	130	seconds
Average burdened wage rate for CSRs (per hour)	\$22.00	
Hours worked per day	8	
CSR Churn Rate	49%	
Reduction in churn rate with AA	4.8%	
Cost to replace a CSR	100%	of wages
Hours worked per week	40	
Number of shifts?	1	
CSR non-talk time (hours per day)	2.00	
% of callers that hang-up and call back again later.	3.00%	
% of calls that hang-up and call back again later.	10%	
Reduction in the % of callers that hang-up and call back again later.	10%	
Cost to license Adaptive Audio	\$294,717	
Additional talk time for caller to vent	5.00	Seconds
% of additional callers that will use self-service if it has Adaptive Audio technology	1.00%	
Per minute toll rate	\$0.020	
Busy-hour calls	17,045	
Reduction in self-service call length with Adaptive Audio Technology	8%	
Enterprise Variables:		
% of callers that are lost potential customers	0.010%	
Monthly Lost revenue from callers that are lost potential customers	\$50	
% of callers that are customers that don't get issue resolved properly	0.0500%	
Reduction in lost customers with Adaptive Audio technology	10%	
Cost to acquire a new customer	\$300	
Amount spent on advertising	\$10,841,751	per year
Reduction in advertising expense	1%	

Obtain results here (see following sheets for additional details):

Reduction in Self-Service Call Length with Adaptive Audio Technology	8%	
Call Center Payback Period	3.108	month
Enterprise Payback Period	2.012	month
Call Center ROI	386.1%	
Enterprise ROI	456.6%	
Cost Reduction in Call Center Operating Expense	\$4,310	per day
	\$94,827	per month
	\$1,137,929	per year
Cost Reduction in Enterprise Operating Expense per year	\$1,345,706	per year
Start number of CSRs	271	
CSRs after adding AA	254	
Reduction in required CSRs	17	
Cost Reduction of CSR labor (venting callers)	\$224,889	per year
Cost Reduction of CSR labor (additional self-service utilization)	\$547,556	per year
Cost Saving Due To CSR Churn Reduction	\$296,206	per year
Telephone Toll Expense Reduction	\$69,279	per year
Savings From Reduction In Lost Customers	\$99,360	per year
Cost of Additional Enterprise Advertising	\$108,418	per year
Total Enterprise Expense Cost Reduction	\$1,345,706	per year
Total Enterprise Operation Expenses	\$44,485,124	per year

Figure 11
Adaptive Audio savings calculations for a large-size Call Center

Call Center Variables:		
Calls per month	10,000,000	
Average length of talk time per call (seconds)	280	
% of calls that are completed by self-service	70%	
Average length of self-service call	110	seconds
Average burdened wage rate for CSRs (per hour)	\$22.00	
Hours worked per day	8	
CSR Churn Rate	49%	
Reduction in churn rate with AA	4.8%	
Cost to replace a CSR	100%	of wages
Hours worked per week	40	
Number of shifts?	1	
CSR non-talk time (hours per day)	2.00	
% of callers that hang-up and call back again later.	3.00%	
% of calls that hang-up and call back again later.	10%	
Reduction in the % of callers that hang-up and call back again later.	10%	
Cost to license Adaptive Audio	\$1,214,570	
Additional talk time for caller to vent	5.00	Seconds
% of additional callers that will use self-service if it has Adaptive Audio technology	1.00%	
Per minute toll rate	\$0.020	
Busy-hour calls	85,227	
Reduction in self-service call length with Adaptive Audio Technology	8%	
Enterprise Variables:		
% of callers that are lost potential customers	0.010%	
Monthly Lost revenue from callers that are lost potential customers	\$50	
% of callers that are customers that don't get issue resolved properly	0.0500%	
Reduction in lost customers with Adaptive Audio technology	10%	
Cost to acquire a new customer	\$300	
Amount spent on advertising	\$54,208,754	per year
Reduction in advertising expense	1%	

Obtain results here (see following sheets for additional details):

Reduction in Self-Service Call Length with Adaptive Audio Technology	8%	
Call Center Payback Period	2.579	month
Enterprise Payback Period	1.666	month
Call Center ROI	465.4%	
Enterprise ROI	550.9%	
Cost Reduction in Call Center Operating Expense	\$21,410	per day
	\$471,017	per month
	\$5,652,204	per year
Cost Reduction in Enterprise Operating Expense per year	\$6,691,091	per year
Start number of CSRs	1,355	
CSRs after adding AA	1,272	
Reduction in required CSRs	83	
Cost Reduction of CSR labor (venting callers)	\$1,124,444	per year
Cost Reduction of CSR labor (additional self-service utilization)	\$2,737,778	per year
Cost Saving Due To CSR Churn Reduction	\$1,481,028	per year
Telephone Toll Expense Reduction	\$308,953	per year
Savings From Reduction In Lost Customers	\$496,800	per year
Cost of Additional Enterprise Advertising	\$542,088	per year
Total Enterprise Expense Cost Reduction	\$6,691,091	per year
Total Enterprise Operation Expenses	\$221,865,619	per year

Conclusion

With Call Centers, Enterprise IT Departments and ASR based hosting centers recognizing the economic benefits of automated calls versus using an agent, the trend is towards longer, more complex and information-rich speech applications. Enabling technologies such as the Web-Centric IVR, speech enabled dialogs, natural Language Understanding and Customer-Directed Dialogs serve to help further this trend.

With this increased caller interaction, leveraging the benefits an Adaptive VUI offers several Direct and Indirect Benefits including Increased Operational Efficiencies, Reduced Operational Costs, Increased Customer Satisfaction and a Very Short and Verifiable ROI and payback period. Achieving these benefits via an easily installable software-only product makes good business sense from the point of view of the Customer and the Enterprise.

About Interactive Digital

Interactive Digital pioneered the implementation of Adaptive Voice Technologies for use in telecommunications. Our focus is on improving IT efficiencies and saving call center and enterprise costs through the use of our Adaptive Technologies. In combination with our partnerships and affiliations with major players in the speech industry, Interactive Digital prides itself on its excellent client service and guaranteed savings for the call center and enterprise.

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Other Patents Pending