Functional Design
Response to Customer X
Requirements Documents,

CUSTOMER X Telephony Upgrades
&
Technical Detailed Design Specification

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Revision 1.0

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Executive Summary

Background:

The purpose of this document is to detail current recommendations by Provider #1 Corporation in response to CUSTOMER X requirements described in two separate documents; "Business Requirements Document for CUSTOMER X Telephony Upgrades", and "RIO Technical Detailed Design Specification". Provider #1 staff greatly appreciates the detailed descriptions that have been identified by CUSTOMER X. Both partners have been exploring the possibilities of using Voice over IP implementations over an extended period and have reached a point in our individual technology deployments where very real possibilities exist. In addition to specific VoIP technical requirements, CUSTOMER X has also defined a wide range of business needs designed to improve overall performance of the call processing components in both IVR and Call Center solutions. These objectives combine to improve both the caller experience and expense models for CUSTOMER X.

Approach:

The two requirements documents generally describe two separate topical areas for our response. (1) Voice over IP technology deployment (based on the "RIO" requirements); and (2) Call Management improvements (from the "Telephony Upgrades" document). The following materials will provide separate sections based on those categories, but there will be some overlap between the two areas in terms of pre-requisites and/or co-dependencies for both technical and procedural elements. In responding to all requirements, we will attempt to keep these distinctions clear. While there are probably many different ways to "meet" some of CUSTOMER X's objectives, Provider #1 will provide our best recommendations to meet these challenges.

Because of the range of dependencies between the technical and business requirements, and the need for consideration and planning for both organizations, we will not try to present firm "schedules" as part of this response; but we will provide logical sequences for implementation consideration that can become part of the detailed planning that we both do going forward. We will also provide our estimates of the difficulty of some of these items that we could jointly use to model implementation durations.

The response also provides estimates of the software and hardware components that would be required to meet the needs defined in the respective requirements sections. Since the requirements documents did not specify locations and volumes at this time (although we recognize where CUSTOMER X may include this later), these estimates will be based on the volumes and locations for CUSTOMER X that Provider #1 interfaces with today.

P.O.C (Proof of Concept):

As described in section 2.3 of the "Telephony Requirements", we are also providing our recommendations for Proof of Concept testing in June 2009. We concur with that document, that; while Provider #1 is making our best recommendations for the overall requirements in both documents provided; the two organizations will be working together to make potential refinements based on the results of P.O.C. testing.

Summary:

Provider #1 appreciates the strong relationship that we have with CUSTOMER X and welcomes this opportunity to work together on determining the “best” possible solutions to improve both the “customer experience” and overall “call management” functionality for you. While our organizations have demonstrated a strong desire to reach for these goals in the past; with both organizations deploying complimentary new technologies, the possibilities for accomplishing these objectives shows more and more promise for the future.

Technical Summary and General Architecture

CUSTOMER X and Provider #1 are continuously engaged in examining new possibilities for call processing and have been focused for some time on methods involving Voice over IP and new approaches to call routing. We have jointly pursued a number of tests and have stayed in close contact as we have respectively deployed new capabilities that now make more specific pursuit of some of the discussed solutions possible. This document is focused on presenting specific recommendations for some of those solutions in response to CUSTOMER X's stated requirements.

From our analysis, we've chosen to classify the CUSTOMER X requirements under two prominent high level headings, "Voice over IP Call Transport", and "Centralized Call Queuing". While there are some overlapping characteristics to both of these categorizations, we are choosing these classifications for several reasons: (1) There are some solutions being suggested that involve both of these headings, but more of the requirements uniquely fit well under one or the other. (2) We received two requirements documents to evaluate and make recommendations from. These documents also overlap to some degree but tend to align to the respective headings as well ("RIO Requirements" → VoIP and "Telephony Upgrades" → Centralized Call Queuing). (3) As we proceed forward and begin to consider the financial parameters of recommended solutions, it may be more practical to look at the benefits related to the two headings in preparing our respective business cases.

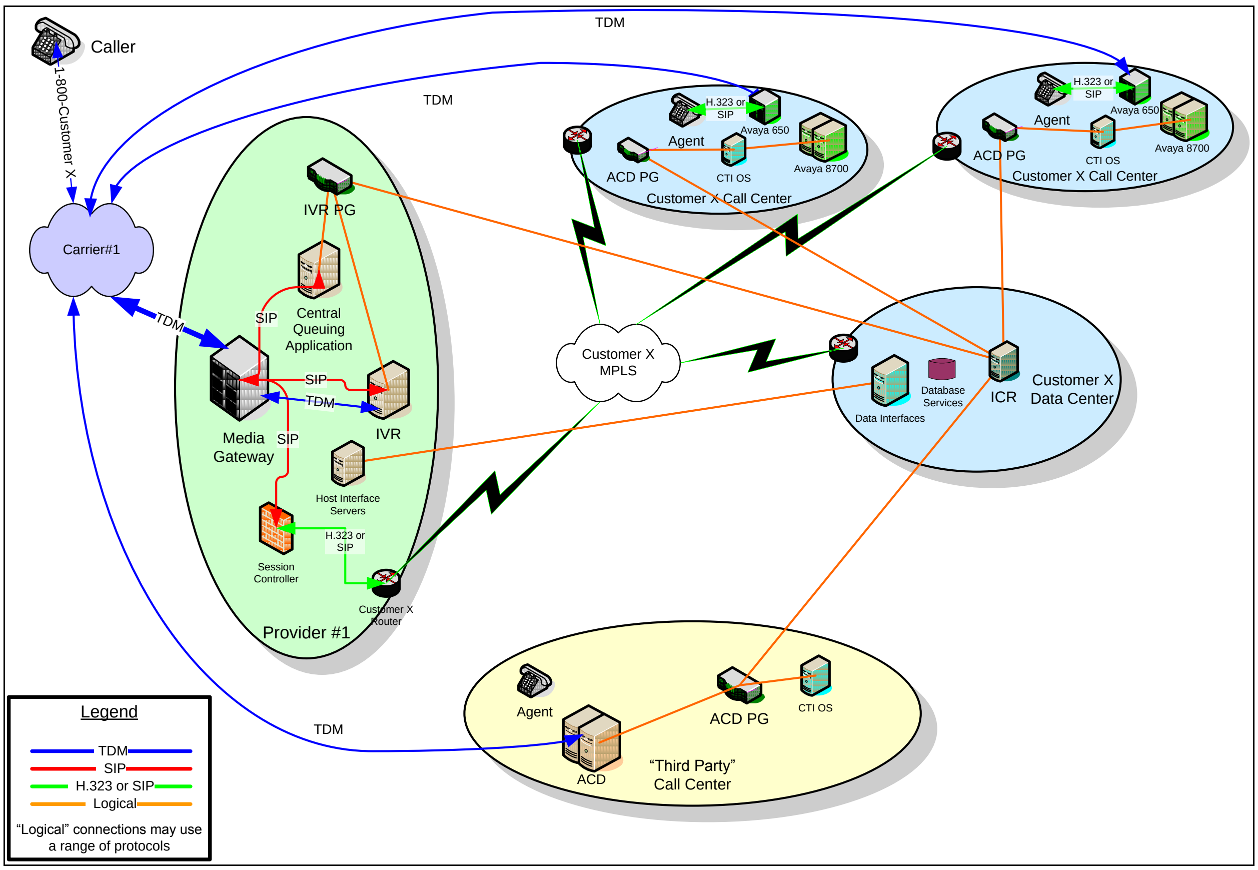
Figure 1, below, illustrates an overall high-level concept for meeting the combined requirements of both VoIP network and the ability to manage a consolidated queue for calls be passed to CUSTOMER X contact centers. It is not intended to represent the details of unique characteristics that may exist at specific locations; but to show the basic functionality of all processing components. We will explain details of specific call flows in greater detail later in this document.

Key Elements Of High-Level Processing Architecture:

1. The image illustrates call delivery into a single Provider #1 location. In practice this solution would be deployed at multiple Provider #1 locations and the physical elements would be redundant.
2. 1-800-CUSTOMER X calls continue to arrive at Provider #1 processing centers and are routed to the existing IVR application systems (to include all data interfaces and speech technology) for processing. These IVR systems may use either TDM or IP (SIP) connections to the Media Gateway. When a call requires transfer to a contact center, the IVR requests routing information from the local ICM Peripheral Gateway (IVR PG).
3. IVR PG requests/receives routing information from the ICR (ICM call controller) and instructs IVR as to call disposition.
4. If the route request indicates that the destination is a contact center on CUSTOMER X's private network (MPLS?), the new architecture would use the Provider #1 Media Gateway to switch the call to VoIP and deliver it to the indicated center. If the destination does not support IP delivery, the Media Gateway can still route calls using CARRIER#1's Transfer Connect feature.

5. If the route request finds no operator available, it should "route" to an alternate IVR application that operates as a "Central Queue". This IVR app will operate under the direction of CUSTOMER X's ICR using Cisco's ICM Service Control Interface that will allow CUSTOMER X control over RAN messaging and caller interface, in addition to Call Routing. (more details discussed in later section).
6. When the ICR determines destination availability, the Central Queue IVR application passes data to ICM and requests the Media Gateway to route the call in the same fashion as the "classic" IVR application.

Figure 1: High-level Concept for IP Call Delivery with Centralized Queuing to Call Centers



Key Opportunities with "New" Architecture:

1. Calls utilizing the "private" network for delivery to call centers could avoid Transfer Connect requirements.
2. Central Queuing application allows CUSTOMER X to establish routing based on skillsets and resources across multiple locations.
3. ICM VRU direction via Service Control Interface for messaging and script execution.
4. Ability to consolidate information regarding overall call handling by maintaining call data associations to ICM Call Key (staying connected to Media Gateway)
5. Allows for "transition" to new technologies versus "conversion".

Additional information on each of these elements is provided in the sections focusing on “RIO Requirements” and “Business Requirements for Telephony Upgrades”.

Response to "RIO Technical Detailed Design Specification (Draft)

Overview

The "Rio Requirements" document provides the key guidance on issues related to the potential implementation of private networking(VoIP) and the continuing operation of existing operational tools for call routing and CTI affecting delivery of caller information to agent desktops. Provider #1 understands the critical nature of ICM and Siebel systems on the operational models that are in place, and that new implementations must not cause any functional disruption of these systems' operation. While some recommendations may present alternatives to current procedures, those potential changes should be capable of being implemented on a "migratory" basis as opposed to requiring any mass change to physical systems or network deployment. With any alternatives chosen, Provider #1 will work closely with CUSTOMER X to ensure that impact is minimized.

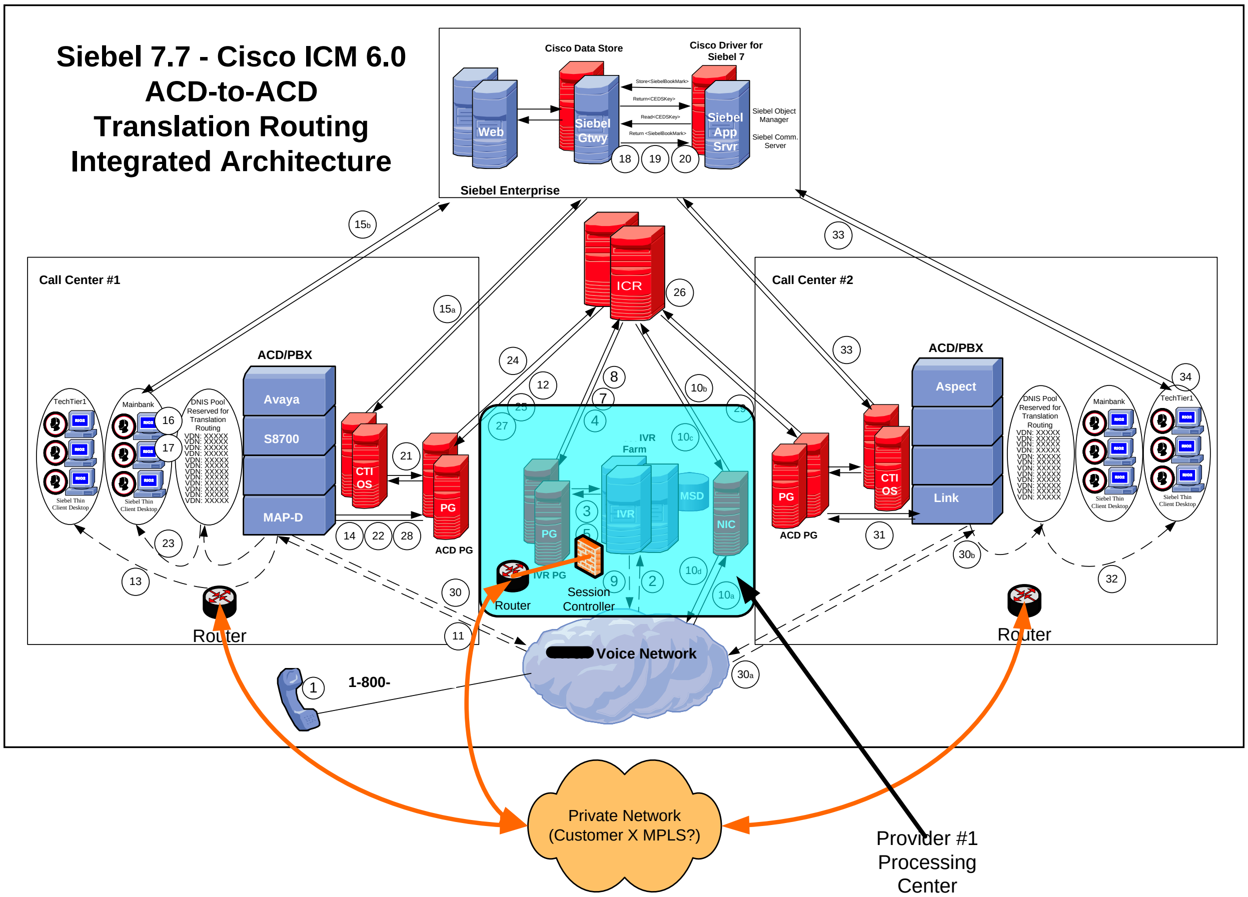
In order to achieve potential cost savings relief from current charges for network transfer of calls (Carrier#1 "Transfer Connect"), it is necessary for the parties to deploy some type of "private" network interconnection scheme to eliminate the requirement to "go back" to the carrier network. Several alternatives exist for such connectivity. These alternatives could include both TDM and IP telephony options, but for purposes of this response, we will be looking at IP for reasons of flexibility in protocol, bandwidth, network integration of voice and data that may contribute to future requirements not currently identified. It is also Provider #1's expectation that CUSTOMER X would want to "own" the network element for overall system management. For these reasons we will address most of our consideration to an expectation of using CUSTOMER X MPLS network connections from Provider #1 processing centers to your preferred destination locations.

Summary of VoIP Recommendation

In order to provide CUSTOMER X opportunities for relief from costs associated with the use of carrier network charges for capabilities such as "network transfer" of calls, and to provide alternatives for technology improvements in the areas of protocol uses, routing options, and data convergence; Provider #1 recommends that both parties pursue detailed design efforts to deploy IP Telephony as the primary interconnections between our locations for call processing.

Figure 1 in the above "Technical Summary" section of this document illustrates, at a high level, the connectivity. We believe that this capability could be implemented with little change to the operational mechanisms that currently exist for Call Routing (ICM) and CTI (Siebel) because the solution would implement IP for the "Voice Path" for calls, but would not alter the physical connections, or primary operation of ICM and CTI elements. As an alternative view, figure 2, below, uses the operational schematic from the "RIO Requirements and "adds" a private network option for call delivery. **Cost savings could be derived from two perspectives: (1) The use of the private network to replace "transfer connect", and (2) the ability to reduce, over time, the number of local trunk connections into call center locations.**

Figure 2: Image from "Rio Requirements" + IP



In this image, existing components stay in place for call processing. All ICM and CTI mechanisms remain. A private network is established across the locations to support the transport of voice calls via IP. Under this model, most of the calls would initially arrive at the Provider #1 IVR locations from the carrier via TDM origination on the PSTN, would "stay" on the Provider #1 platform, and be "switched" to VoIP for delivery to Call Centers as required. Since Provider #1 will support the "ingress" for the full call duration instead of transferring the call back to the network; we will add new CARRIER#1 inbound capacity to our media gateways. Additional discussion of the private network is described below in the "VoIP Solution Components" section. The following paragraphs outline why there is limited impact to the ICM/CTI operational paradigms with the VoIP solution.

Media Gateway / ICM Interface & Operation

While we generally refer to our Sonus Media Gateway at Provider #1 as a "single" entity, the Sonus switching solution is a "carrier-class" CLASS 4 switching platform that actually can consist of a number of physical devices/servers depending on the scope of the switching environment. For purposes of explaining our implementation and interaction in our proposed solutions, we will discuss the Sonus GSX 9000 and the Sonus PSX.

The Sonus GSX 9000 is a high-density, open-services, media gateway providing a wide range of protocol switching capabilities and scalable to millions of ports, and thousands of transactions per second. This is the physical connectivity device for our network interfaces for both circuit-switched, or TDM (DS1, DS3, E1, OC3/STM-1) or IP (Ethernet, Gigabit Ethernet) technologies. Provider #1 has implemented this switching architecture as a front-end to our self-service processing environment to allow us to migrate to, and blend multiple protocol solutions for our clients based on their needs, while minimizing the impact of such potentially major change on actual processing systems..

Sonus' PSX Peripheral Routing Server provides software-based routing control over the GSX 9000. Designed for Carrier implementations, the PSX transaction processing (routing transactions per second) is only limited by the scale of the systems that it runs on and the Provider #1 platform can handle hundreds per second. In carrier implementations, it is the PSX that handles extensive services for "least cost routing" solutions that may require analysis of numerous paths to a destination. In our comparatively simple routing decisions, the PSX provides more than sufficient system intelligence and performance.

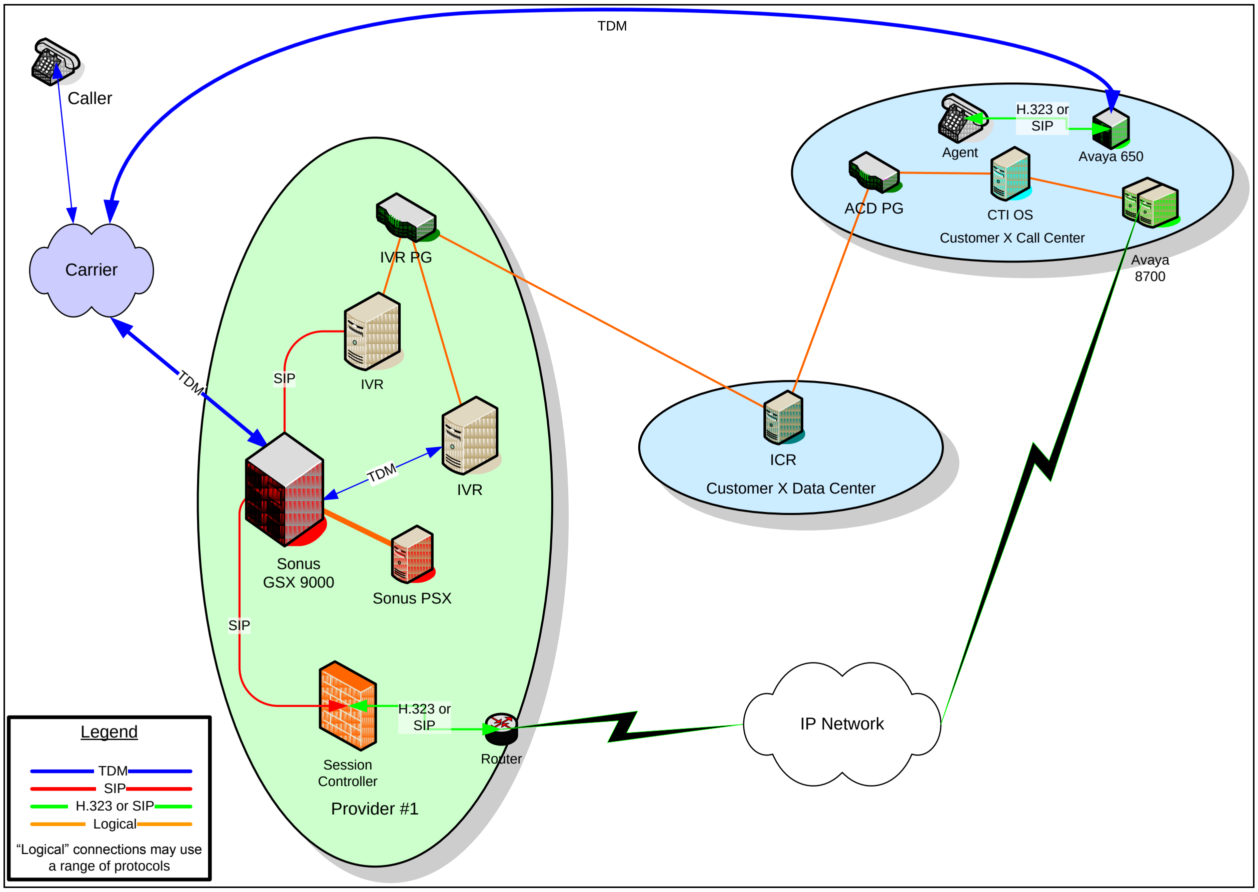
While the Media Gateway architecture also supports other components depending on a number of factors, we will limit the necessary discussion for these projects to the GSX and PSX. Most of our documentation with simply refer to the "Media Gateway" as a single moniker for these components. The figure below will be used to illustrate the narrative discussion of ICM and Media Gateway function.

Call Processing

All calls to be handled on the Media Gateway are subject to specific routing or default terminations. All calls arriving on the GSX 9000 are examined for DNIS and the PSX tells the GSX where to deliver the call. For the majority of our inbound traffic, calls will be delivered to either TDM or IP IVR systems for call processing. It is important to note that the nature of this "open services" media gateway is to allow different types of protocols and physical connectivity. The primary function of media gateway is to resolve the differences and support connecting different methods together. The same routing resolution operates on "outbound" call requests as well. For example, if an IVR system needs to transfer a call, it will "send" a transfer request to the GSX. The GSX will pass DNIS information to the PSX for resolution. In the case of a transfer to a VoIP-enable destination, the PSX will return the IP address of the destination and the GSX will structure the appropriate H.323 or SIP "invite" message to initiate the call. That message will still include the "dialed number" that the PSX used to look up the route that may be in a form like 8yy-999-9999. The receiving system can then make decisions based on that "standard" formatted number.

Using the following figure, let's look at a couple examples:

Figure 3: Media Gateway Interaction with ICM/CTI



Example #1:

a. A call is delivered to the Provider #1 processing center from CARRIER#1 using a TDM trunk. It is delivered to an IVR system and the script determines that it must transfer the call to an operator center.
b. We are using ICM as a call routing and CTI data pass mechanism. The IVR sends a "Request Route Key" the local IVR Peripheral Gateway and after a response to that request, it sends a "Route Request" along with any necessary data to be provided at the receiving location when the call "lands".
c. The ICR call routing controller looks at the enterprise information and selects a "destination" that is a TDM trunk over the PSTN. It associates a Dialed Number with that location (to subsequently match call data to) and returns that destination number (8yy-222-3333) to the IVR.
d. The IVR signals the media gateway that it has a transfer request and passes the destination DNIS to it. The Sonus GSX checks the PSX for routing information and is "told" to use an outbound TDM trunk to deliver the call to and to dial the same number, 8yy-222-3333.
e. The GSX will use a standard CARRIER#1 "transfer connect" control mechanism to pass the call to the destination and that destination will use the DNIS to check with it's local ICM components to retrieve data from the IVR process and populate an operator screen.

This is, obviously, a familiar scenario to us. What is important is that the PSX plays a key role in this implementation and actually determines the final number to be "dialed" to send the transfer to. Historically, the PSX would just return the same number for TDM calls, but what if we want a different outcome. The PSX allows Provider #1 to provide alternative "provisioning" for number processing and call termination. By simply providing alternative destination and protocol at the PSX, we can change to a VoIP protocol.

Example #2:

Steps a through c above operate in the same way, but this time the destination number, 8yy-222-3333, has been identified in the Provider #1 PSX to use a VoIP network and a specific IP address to send the call to…..

d. The IVR signals the media gateway that it has a transfer request and passes the destination DNIS to it. The Sonus GSX checks the PSX for routing information and is "told" to use a VoIP route and send the call to IP address 111.222.333.444 with a SIP "Invite" request. The "dialed number" field in the Invite will still reflect 8yy-222-3333
e. The GSX structures the "Invite" message and sends it to the indicated IP address which should be a proxy server for the destination. Processing will continue at the destination using the DNIS field in the "Invite" to check with it's local ICM components to retrieve data from the IVR process and populate an operator screen.

Media Gateway / ICM Processing Summary:

As this indicates, processing with ICM, is in effect unchanged as ICM is never involved in the actual "voice path" for a call. Full implementation will require "provisioning" with the Provider #1 PSX system to indicate how Provider #1 should treat specific destination numbers. While this may, initially, add a step to our overall handling of a number for routes that we want to use IP, it also offers additional options in the route selection that may be considered without change to the ICM operation. Provider #1 currently does the "provisioning" on the PSX, but may consider design options that would allow customers to facilitate their own number management on the PSX in the future.

Appendix A in this response illustrates detailed call flow interactions and signaling processes between the components involved in the call processing for 4 primary variations of call types. The "RIO Requirements" outlines a significant number of call processes that all involve slight change or combination of the functions in these four types or require alternative processing suggestions. Provider #1 will provide additional documentation to include the detail for all of these circumstances in a separate document.

VoIP Solution Components

This section summarizes the key elements required to support the deployment of VoIP between Provider #1 processing locations and CUSTOMER X call centers with the same operations models for ICM call routing, and CTI coordination.

Component	"Owner"	Operation
CARRIER#1 Carrier Ingress	Provider #1 CUSTOMER X	Under the VoIP processing model, Provider #1 would expand CARRIER#1 ingress at processing centers to be able to meet simultaneous call requirements into the call centers. Calls would be "switched" on the Provider #1 media gateway to IP protocols for delivery over the private network to call center locations. CUSTOMER X would continue to have ingress trunks to call center locations to support traffic that it determined was not routed through the IVR systems or other local administrative traffic. Capacity supporting current toll-free number termination should be reduced over time as it is moved to the IP network
Sonus Media Gateway	Provider #1	Provides termination for ingress ports to carrier for toll-free number termination on the PSTN. Supports interface to deliver those calls to IP telephony protocols and destinations. Has integrated call routing database supporting destination/protocol provisioning for control of call delivery. This call routing database provides the associations for CUSTOMER X DNIS to indicate TDM (route is a Toll Free Number) or IP (route is a Call Center IP address)
IVR Systems	Provider #1	Provider #1 custom Voice Self Service platform including all existing data interfaces, ICM communications, and Speech technology integration. There is no expected change to the operations of IVR for the VoIP paradigm. (other functional changes will be discussed in the "Telephony Upgrades" section of this document.
Routers	Provider #1 & CUSTOMER X	Each party would be expected to provide "edge" routing capabilities at their respective termination points. Router configurations will vary depending on final determination of voice transmission protocols & formats (see "Initial Sizing Estimates" section below)
Session Border Controller	Provider #1 CUSTOMER X(?)	Session Border Controllers(SBC) are specialty "firewall" products designed to support IP telephony traffic. It is up to the respective partners to make their own Security Requirements determination as to the applicability in each installation. Provider #1 security paradigm requires the use of SBCs when connecting to outside networks and can assist CUSTOMER X in evaluating available solutions and making recommendations. Some SBCs also can provide protocol conversion facilities that may reduce/simplify the operation of other network components.
Cisco ICM products - ICR (call routing controller) - Peripheral gateways (IVR & ACD)	CUSTOMER X	ICM components operated/scripted by CUSTOMER X should function in the same fashion for VoIP over IP call delivery (see above discussion on Media Gateway / ICM interaction). Provider #1 will need "destination" IP routing information to be provided that will make the association between DNIS and Call Center IP addresses.
Seibel CTI OS Systems	CUSTOMER X	There should be no changes in the CTI systems related to VoIP call delivery.
Avaya ACD	CUSTOMER X	The Avaya ACDs will require configuration changes to identify

**For VoIP implementation the only interface would be expected on the 8700 system. The Avaya 650s would only be required if private network connections were used with TDM transport		"IP trunking" that will accept traffic from the private IP network, operational protocols (SIP, H.323), and voice packet characteristics (codecs, sample rates, etc.). Provider #1 supports the Avaya configurations in our operational centers and lab environments and will work with CUSTOMER X to make the most appropriate final recommendations and selections of these criteria for the CUSTOMER X environment.
Agent Resources (handsets, softphones, desktop systems, etc.)	CUSTOMER X	There should be no changes in the agent systems required for VoIP call delivery to call center locations.

“VoIP” - Initial Volume Estimates and Considerations

As mentioned previously, since the requirements at this time did not outline specific volumes or capacity the following information is provided for discussion purposes and is based on Provider #1’s current understanding of the calls that we are assisting with. As we look at the details for a full implementation together we have developed a wide range of IP Telephony analysis tools that may assist us in those conversations.

Media Gateway impact

In order to support the maintaining of calls to agents, in progress, to avoid “transfer connect” charges; Provider #1 will add ingress trunks to our media gateways. In analyzing our traffic and transfer rates, we estimate the need to support up to 10,000 simultaneous calls to operators. Our initial considerations have indicated that we could meet this objective by adding approximately 2000 ports of ingress in each of three processing centers and obtaining the balance of the 10,000 from our existing shared resources. This will require expansion of the Sonus GSX and additional to CARRIER#1.

Private Network Design and impact

The actual design of the private network will require careful planning by both parties. While we may discuss a wide range of issues in solving all the challenges outlined in this document, the private network configuration will have the most impact on total costs for the parties, and performance and call quality for callers and agents. Careful determination of protocol (SIP, H.323) and compression options (codecs: G.711, G.729, etc.) are required. We expect the Proof of Concept testing to deliver results that both parties will use in these determinations.

Based on our current data and the need to support up to 10,000 simultaneous calls, we believe that there is positive opportunity for savings. However, actual figures can only be calculated based on our planning discussions. The nature of fixed costs with IP telephony versus variable with TDM models can produce a wide range of results. For instance, an OC12 circuit could support around 6000 simultaneous calls using a typical sample size and G.711 compression, but the same circuit, same cost could support over 16,000 running G.729 compression. While G.729 produces “less” quality, technically, than G.711; it also is, statistically as good or better than the compression used on most of our mobile phones and cannot be quickly dismissed.

“Carrier” Private Network

Provider #1 has been in close contact with CARRIER#1 on their new VTN network offerings that allow the use of IP Telephony over carrier provided “shared” network elements for Virtual Private Networking. We are aware that CUSTOMER X has had “early” briefings on the technology as well. This will present an additional possibility to establish the connections between locations that we require for the solutions. Additional due diligence is required with regard to whether or not the full scope of signaling control would be available to us over this solution.

Response to Telephony Upgrades Requirements

Overview

CUSTOMER X has done an excellent job articulating the scope of challenges faced in managing telephony infrastructure for call processing. As indicated previously, our analysis indicates that the key to delivering a majority of the needs described in this requirements document is based on the need to provide a "Common Queuing" mechanism within the Provider #1 processing environment that can support the routing of calls to call centers based on expanded delivery criteria that will optimize the use of both human and material resources for CUSTOMER X. The following sections discuss how that process can work and some of the discussion points needed in detailed design planning.

Summary of "Common Queuing" Recommendations

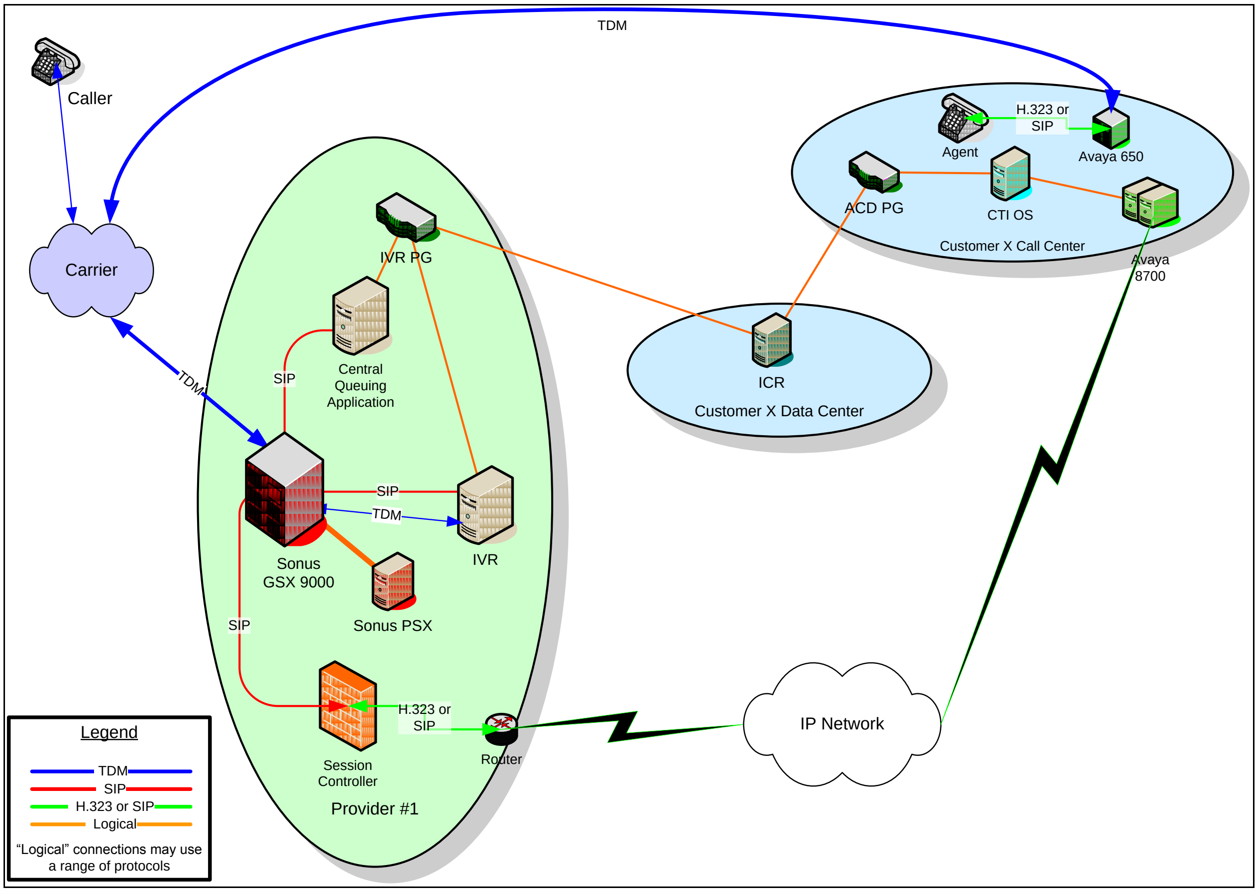
In order to provide a mechanism to provide a "common queue" and associated caller interface objectives to transfer calls to CUSTOMER X call centers, Provider #1 is recommending two substantial changes to the current paradigms that we respectively operate under. To provide this advancement will require configuration and operational changes by both parties**. (1) Provider #1 will introduce a "secondary" IVR platform that would be dedicated to providing the queuing functions.** This platform will be based on our expanding VXML processing environment and is capable of interfacing with Cisco ICM using an alternate ICM control option called "Service Control Interface"(SCI). With SCI, the CUSTOMER X call routing server, ICR, can control actual VRU elements such as messaging and caller interface to meet the objectives described in the requirements. **(2) CUSTOMER X would need to change the control mechanism for the Provider #1 IVR-PG directing the Common Queuing IVRs to use the Service Control Interface.** This involves configuration changes in the particular ICR to PG relationship and the use of new scripting functions for the interface. It does not require a change in the ICR system itself. There are some additional considerations that will be discussed in the "Components" section below.

Call Processing with "Central Queuing"

Figure 4, below, illustrates the components involved in the delivery of Common Queuing. In this solution, calls arriving at Provider #1 processing locations are initially handled in the same fashion as today and are routed to the existing IVR systems with full support for current data interfaces and Speech technologies. When IVR treatment is complete, and/or a business rule dictates transfer to a call center, the IVR will send the usual request to the local IVR-PG. If the ICR immediately has an operator then it will send the usual response with destination to the PG and the IVR will request that the media gateway transfer the call. If the ICR has no operator available then it will return a destination of "Queue" (actual value will be defined in implementation) and the IVR will request the media gateway to "send" the call to the alternate IVR platform. Appropriate call data will be passed to the queuing application for association with the call when an operator is available. The IVR will interface with the IVR-PG and ICR to establish a service control session and the ICR will direct IVR events such as messaging, and/or additional caller interaction. When the ICR determines that it has an operator to transfer to, it signals the transfer information to the IVR, which requests the media gateway to switch the call based on the provisioning associated with the DNIS that it has received.

Using the following figure, let's look at an example:

Figure 4: Call Processing – Central Queueing Components



Example #1:

a. A call is delivered to the Provider #1 processing center from CARRIER#1 using a TDM trunk. It is delivered to an IVR system and the script determines that it must transfer the call to an operator center.
b. We are using ICM as a call routing and CTI data pass mechanism. The IVR sends a "Request Route Key" the local IVR Peripheral Gateway and after a response to that request, it sends a "Route Request" along with any necessary data to be provided at the receiving location when the call "lands".
c. In this example, the ICR determines that no operator is available and returns a destination indicating the Common Queuing platform.
d. The IVR sends a "transfer" request to the Media Gateway and the gateway recognizes the destination to be the "alternate" IVR systems. The original IVR will also provide Call Route Key information and call data to the alternate platform for subsequent CTI needs.

e. The Common Queuing IVR initiates a "new call" with the ICR using the PG Service Control Interface and the ICR takes control of the actions in the IVR processing (messaging, caller interaction).
f. When the ICR determines that it has an operator available it sends a request to the IVR to connect the call and provides the usual DNIS and other required information.
g. As discussed in the "Rio Requirements" response, let's assume that this example will use a IP route to the call center. The IVR signals the media gateway that it has a transfer request and passes the destination DNIS to it. The Sonus GSX checks the PSX for routing information and is "told" to use a VoIP route and send the call to IP address 111.222.333.444 with a SIP "Invite" request. The "dialed number" field in the Invite will still reflect 8yy-222-3333
h. The GSX structures the "Invite" message and sends it to the indicated IP address which should be a proxy server for the destination. Processing will continue at the destination using the DNIS field in the "Invite" to check with it's local ICM components to retrieve data from the IVR process and populate an operator screen.

Common Queuing Processing Summary

The Common Queuing platform is the key to many of the requirements laid out in the "Telephony Upgrades" document. It does require new development by Provider #1 and procedural changes with ICM for CUSTOMER X. Detailed planning is necessary for both parties for a smooth transition, but this solution allows for "migration" to the new solution on a number by number or skill by skill basis. While CUSTOMER X must make changes, we believe that similar adjustments to operations will be required for any solution to meet the stated requirements.

Below we will examine the "Major Features" requirements on a point by point basis and also summarize the necessary components for "Telephony Upgrades".

Responses to Section 2.2 – Major Features:

*2.2.1 – Consistent Customer Experience
Objectives for this requirement are essentially met with the establishment of the Common Queuing alternative IVR system that will operate under CUSTOMER X's control using scripting on the ICR. Provider #1 supplies mechanisms for supporting rapid changes for actual message playback audio content that may also be used to allow CUSTOMER X a high level of control for the overall user experience.

*2.2.2 – Targeted Messaging
Provider #1 currently supports varying degrees of message playback based on customer information retrieved from CUSTOMER X host systems. Additional enhancements are part of our ongoing upgrade processes and would be applicable to functions available on the Common Queuing platform as well

*2.2.3 – Central Queue

Our recommended solution for Central Queuing is described above and based on this solution CUSTOMER Xs ICM implementation not only has visibility, but control of the process. Home agent and alternate call center functionality would also be available based on CUSTOMER X's ICM awareness of those locations, not based on any particular function within the Common Queuing solution itself.

*2.2.4 – Skills Based Routing
Provider #1 cannot currently identify a role in providing this functionality beyond our ability to respond to CUSTOMER X's ICM processing requirements since ICM is the authoritative mechanism in the system. We believe that the capabilities described elsewhere in this response will allow us to do that. As always, we are open to discussing alternate paradigms that may benefit CUSTOMER X in accomplishing this objective.

*2.2.5 – Self Service in Queue
As a future consideration, Provider #1 is interested in pursuing more detail in just how CUSTOMER X would like to provide this type of functionality. We believe that the basic ability to transfer a caller back into generic IVR services could be relatively straightforward. To do so, returning them to the same spot in the IVR that they may have previously left, or to return them to the IVR and maintain a queue position at the same time will require careful coordination with our call data mechanism to ensure that any final transfer to an agent presented appropriate CTI values to them.

*2.2.6 – Reduced Transfer Connect Expense
This is the primary subject of the "Response to Rio Requirement" section above in this document. It is our understanding that our cooperative analysis identifying details for a private network solution must produce a significant lower net cost for the required facilities of both parties than current expenses for Transfer Connect.

*2.2.7 – End to End Call Trace
Provider #1 has previously provided analysis on a number of possibilities on coordinating CUSTOMER X call data with Provider #1 maintained information. It has been determined that coordination based on Route Call Key information is possible and the following objective can be met by storing that "key" data at different points. In addition, the establishment of both Common Queuing and Private Network transport to call centers enables Provider #1 to provide additional "check and balance" points in the process of retaining call information.

1. *The ability to pull all legs of any call including all parking durations*
2. *Provide a single "key" to allow queries to retrieve all call segment data*
3. *Ability to trace using Dialed number, ANI, Date, Time, Destination, Account Number, Call Variables or combinations of these.*
4. *The end to end report should include any and all transfers no matter which transfer method is used.*
5. *Provide the origin of where calls are transferred from.*
6. *Provide the ability to report on all messages played to any caller.*
7. *Disconnect direction, did the caller disconnect, did the platform disconnect, did an agent disconnect etc.*
8. *Differentiate between unique inbound calls vs. transfers.*
9. *Tracking on correct CT & Skill, determine if the call was miss routed.*

*2.2.8 – Central Queue Visibility

Based on our Central Queuing solution recommendation, CUSTOMER X would have full control of the processes running on the IVR systems on the platform. Provider #1 can work with CUSTOMER X to consider additional interfaces that may improve the overall manageability of the applications. We would be happy to discuss interface to the described "DRIVr" tool in more detail.

*2.2.9 – Reporting
Provider #1 support a wide range of analytics across the many applications that we have developed. We can respond to each unique requirement as part of follow-on phases for detailed design.

*2.2.10 – Transfer Data
It is our objective to maintain unique call identification for every interaction and to be able to associate that information with Route Call Keys for all calls transferred (whether TDM or VoIP) to CUSTOMER X agent locations. With our joint background of building a significant number methods for data exchange, we are confident that we can meet new requirements that may be developed based on CUSTOMER X's new approaches to skills and resource management.

*2.2.11 – Segmentation
Provider #1 can work with CUSTOMER X to define call processing based on call or caller characteristics. We already support features like function suppression based on account delinquency and can extend similar logic based on any additional data that CUSTOMER X would like us to work with.

*2.2.12 – DNIS Reduction
As described in above sections, the private network solution that we suggest will operate based on CUSTOMER X defined destinations and becomes a "reference" number in the media gateway as opposed to a specific "at cost" toll-free number. For IP termination Provider #1 and CUSTOMER X could construct our own unique "DNIS" usage and simplify current procedures.

*2.2.13 – Non-IVR Calls
The Common Queuing technology described can also function as an "initial" destination for call processing as opposed to current IVR solution. We would require functional requirements and traffic analysis to be able to respond to any particular implementation need.

*2.2.14 – Quality of Service (QOS)
Provider #1 Corporation has been utilizing VoIP protocols for several years and has established solutions that take a variety of network challenges into account. We can assist CUSTOMER X in developing the most appropriate solution for the networking solution that we jointly identify in detailed design.

*2.2.15 – Default Routing
Provider #1 can work with CUSTOMER X to define alternative criteria for routing decisions and related timing mechanisms for those decisions. As described earlier, out media gateway routing database provides a new level of "network" routing capability that has yet to be exploited as well.

*2.2.16 – Other Considerations

Provider #1 is in agreement with the following defined considerations and believes that we meet them with the described solutions:

1. *The platform be built on a proven technology base.*
2. *All telephony partners need to approve and agree to the overall design.*
3. *If the platform is of a new design, significant testing and usage in a live environment is considered mandatory prior to acceptance of the proposed solution.*
4. *Elimination of DTMF and in-band tones*
5. *All pre-existing features and functionality be preserved as they work today.*

"Common Queuing" Solution Components

This section summarizes the key elements required to support the deployment of a Common Queuing platform between Provider #1 processing locations and CUSTOMER X call centers with the same operations models for ICM call routing, and CTI coordination. Several of these elements are common with the "VoIP Solution Components", but are restated here as well.

Component	"Owner"	Operation
CARRIER#1 Carrier Ingress	Provider #1 CUSTOMER X	Under the VoIP processing model, Provider #1 would expand CARRIER#1 ingress at processing centers to be able to meet simultaneous call requirements into the call centers. Calls would be "switched" on the Provider #1 media gateway to IP protocols for delivery over the private network to call center locations. CUSTOMER X would continue to have ingress trunks to call center locations to support traffic that it determined was not routed through the IVR systems or other local administrative traffic. Capacity supporting current toll-free number termination should be reduced over time as it is moved to the IP network
Sonus Media Gateway	Provider #1	Provides termination for ingress ports to carrier for toll-free number termination on the PSTN. Supports interface to deliver those calls to IP telephony protocols and destinations. Has integrated call routing database supporting destination/protocol provisioning for control of call delivery. This call routing database provides the associations for CUSTOMER X DNIS to indicate TDM (route is a Toll Free Number) or IP (route is a Call Center IP address)
IVR Systems	Provider #1	Provider #1 custom Voice Self Service platform including all existing data interfaces, ICM communications, and Speech technology integration.
"Alternate" IVR platform (for Common Queuing application)	Provider #1	As an alternative IVR processing platform, Provider #1 has a significant implementation of Genesys "VoiceGenie" VXML processing capability. This is in response to some customers that require VXML. It does not support many of the custom features that our "classic" voice self service systems do, but it does have native support for Cisco's ICM Service Control Interface. It will provide a proven high performance application platform for the more simple "queuing" functions required while a caller is waiting for an operator.
Cisco ICM products - ICR (call routing controller) - Peripheral gateways (IVR & ACD)	CUSTOMER X	ICM components operated/scripted by CUSTOMER X should function in the same fashion for VoIP over IP call delivery (see above discussion on Media Gateway / ICM interaction). Provider #1 will need "destination" IP routing information to be provided that will make the association between DNIS and Call Center IP

		addresses. There are elements of deployment for the ICM Service Control Interface that require careful planning for both parties. We have gotten some conflicting information from Cisco as to their ability to support both Call Routing Interface and SCI on the same PG at the same time. As contingencies, we would look at supporting either separate PGs for regular IVR (using CRI) and Queuing IVR (using SCI), or Provider #1 would go ahead and build an interface for SCI to the existing IVR systems. We also will work with CUSTOMER X closely on ensuring a smooth implementation of any scripting changes on the ICR.
Seibel CTI OS Systems	CUSTOMER X	There should be no changes in the CTI systems
Avaya ACD	CUSTOMER X	There should be no changes based on the Central Queuing application.
Agent Resources (handsets, softphones, desktop systems, etc.)	CUSTOMER X	There should be no changes based on the Central Queuing application.

Proof Of Concept Recommendations

Based on the information described the “Telephony Upgrade” requirements, Provider #1 is in agreement to pursue the planning and completion of a Proof Of Concept test that addresses the elements described in Section 2.3 – Assumptions & Dependencies:

- Test functionality using compatible signaling of H.323 and SIP
- Record voice conversation for later analysis.
- Analyze voice quality and make recommendations using industry-accepted methods, such as MOS.
- Test G.711 G.723 and G.729 voice codecs
- Test codecs using different sample rates, packet framing using various payload options, and test compression, such as cRTP
- Test efficacy of Voice Activity Detection
- Provide appropriate DSCP bit marking in voice and signaling packets as required by ATT and Verizon

We believe that we may be able to provide separate documentation and demonstrations related to these specific items that mitigate some of them or quickly define alternate criteria. Provider #1 currently has a number of these basic functions in operation in our lab using the same Media Gateway and Avaya components that will be needed for testing and any production planning.

Provider #1 will provide a separate more comprehensive “Proof Of Concept Recommendations” document to serve as basis for our joint P.O.C. planning.

Recommendations Summary

Provider #1 appreciates the opportunity to evaluate and respond to the items in the "Business Requirements Document for CUSTOMER X Telephony Upgrades", and "RIO Technical Detailed Design Specification". We believe the solutions that we have generally described for VoIP connectivity between our enterprises and for a Common Queuing solution,meet a majority of the needs that you describe within the documents or establish a foundation for practical pursuit of some of the more difficult objectives.

When we first began to deploy IP telephony capabilities, our objective was to prepare to meet our customers' timelines as they required for their business. Our Sonus Media Gateway platform has performed extremely well as we have begun to see those requirements from out clients, and we are confident of our ability to deliver the solution that we've described using that capability.

While the "Common Queuing" requirements presents and new opportunity to us to match your needs, our recommendation is based on sound technologies that have a proven track record. It is always our goal to create stable secure operating environments by assembling those proven products effectively.

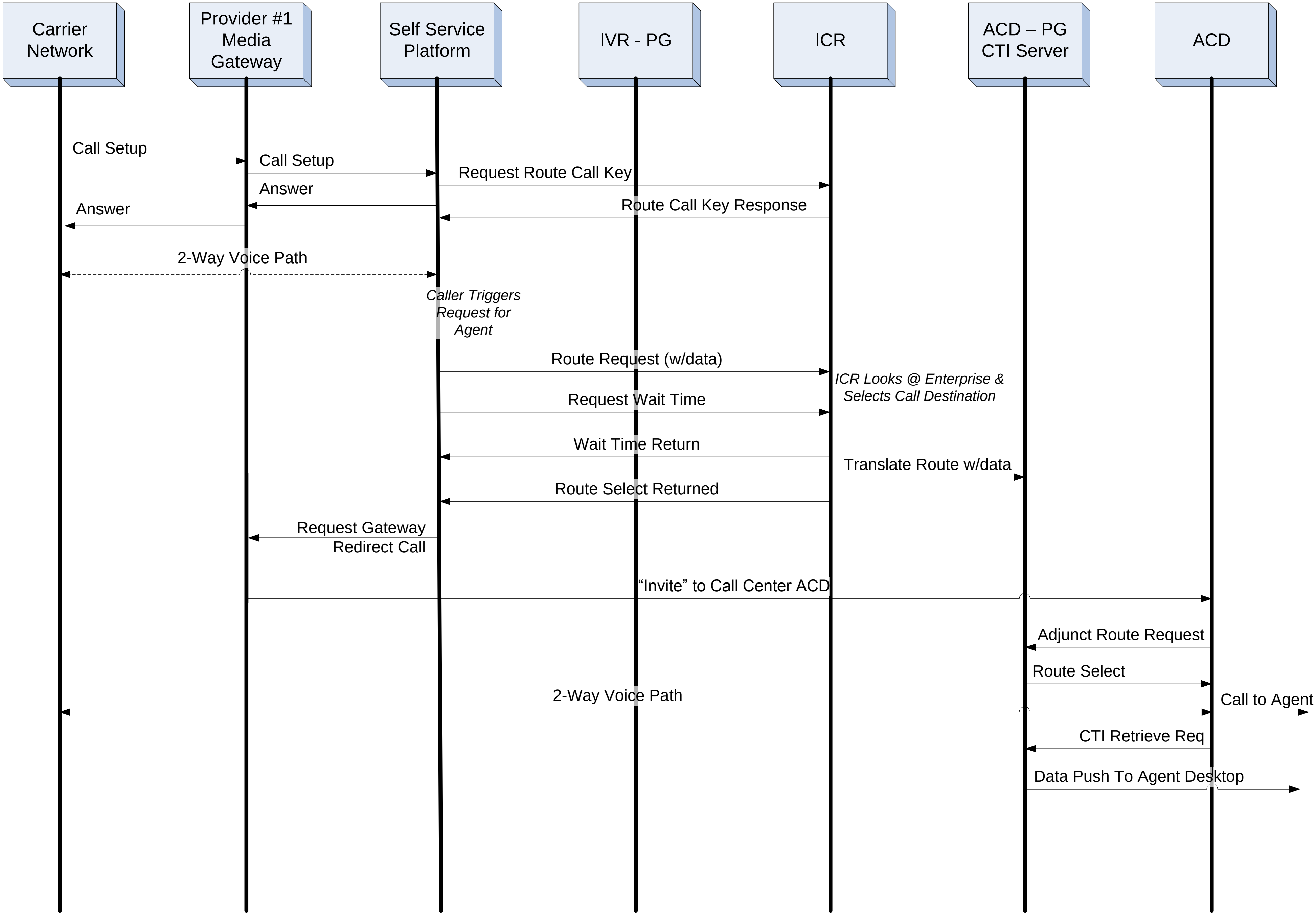
While we are dedicated to assisting our clients in providing the most effective solutions in "self-service", we recognize that the capabilities of those solutions must integrate with the human interfaces that ultimately drive our businesses as well.

We look forward to continuing our detailed design and planning along with our joint Proof Of Concept testing in the very near future.

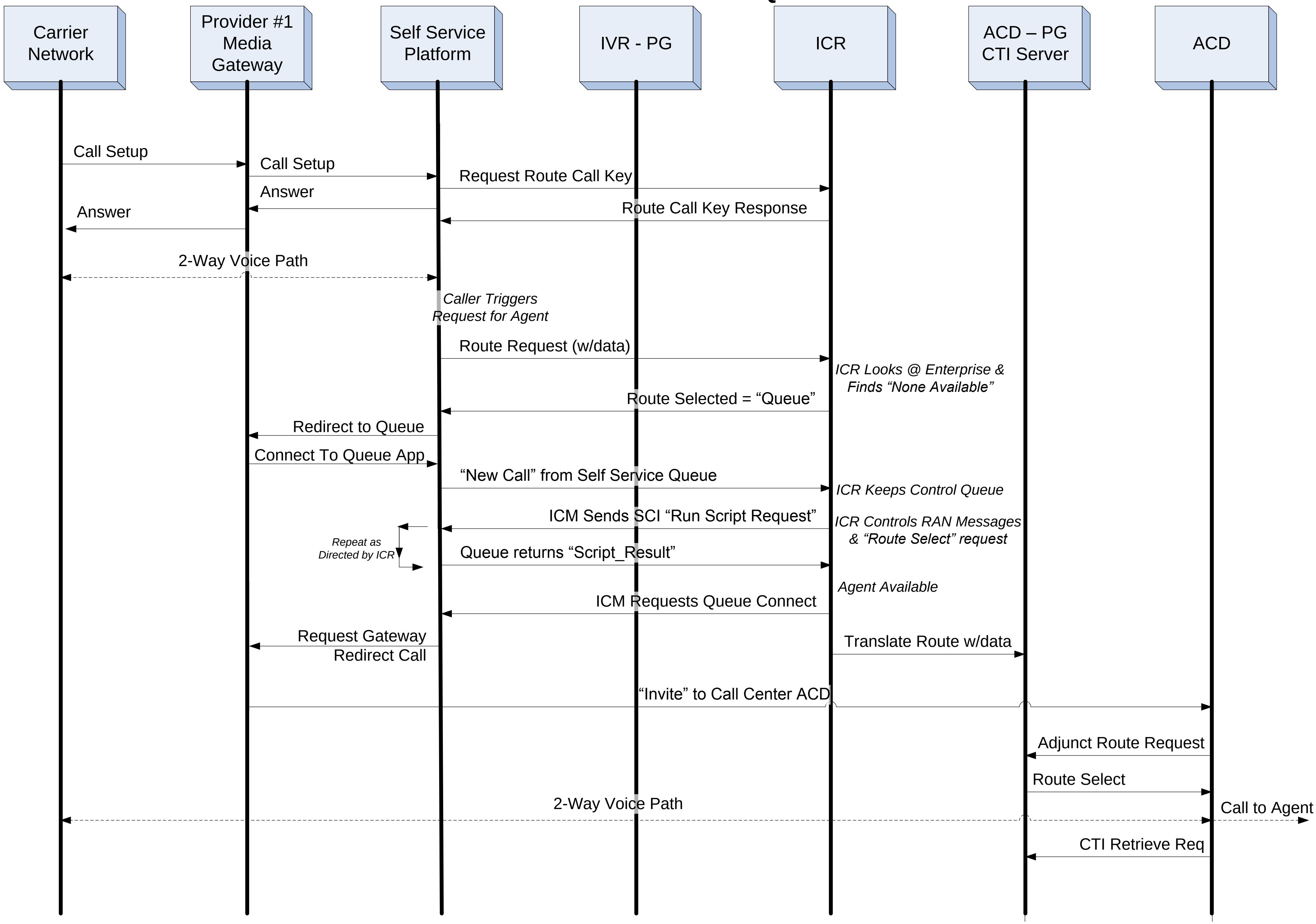
Appendix A – Call Sequence Diagrams

The following diagrams represent our determination of the major functions of the "Recommended Test Scenarios" from the "Rio Requirements document. Provider #1 will provide a separate document that addresses these test scenarios individually and identifies the variations from these sequences or recommended alternative solutions.

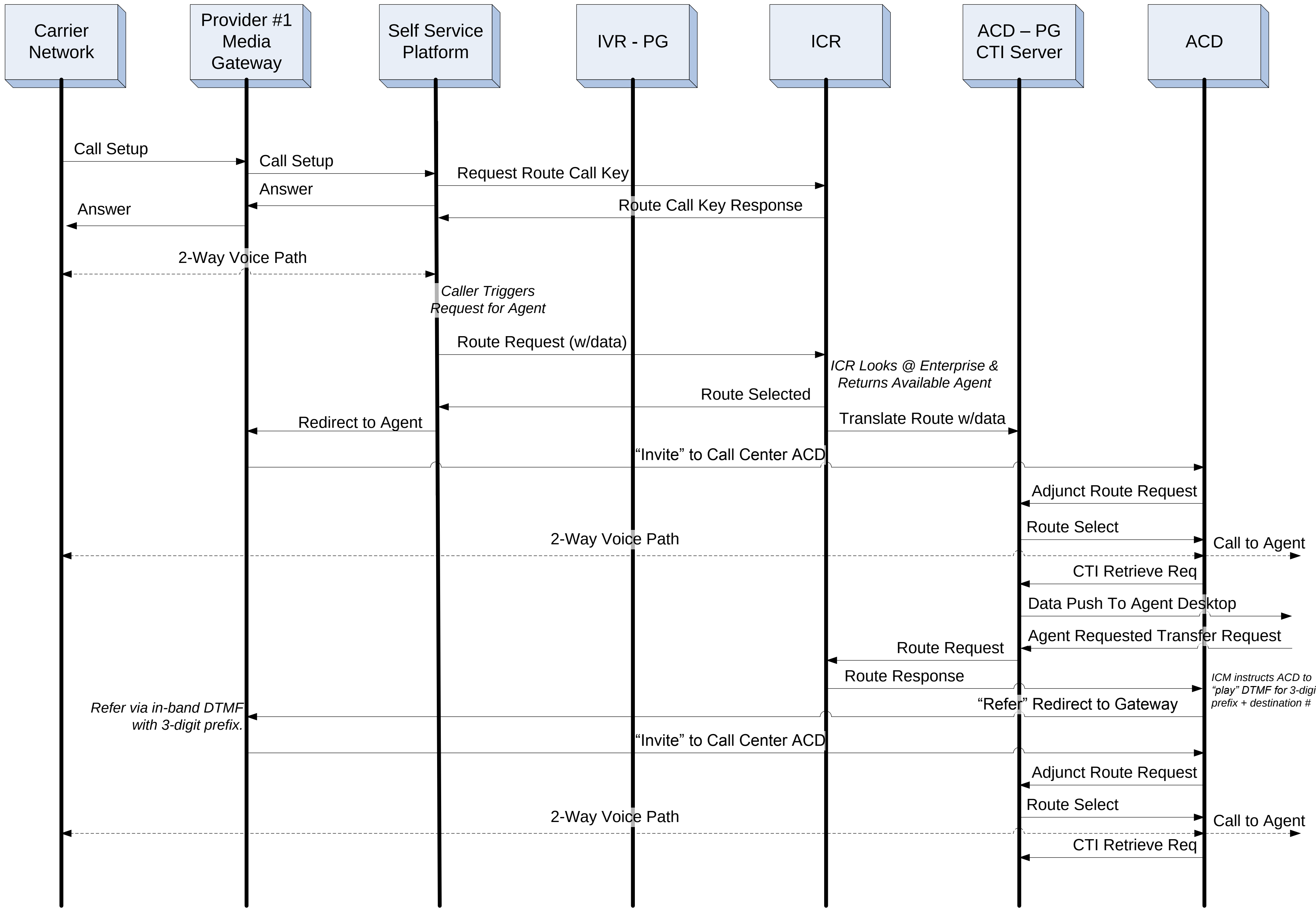
Route Call To/From IVR



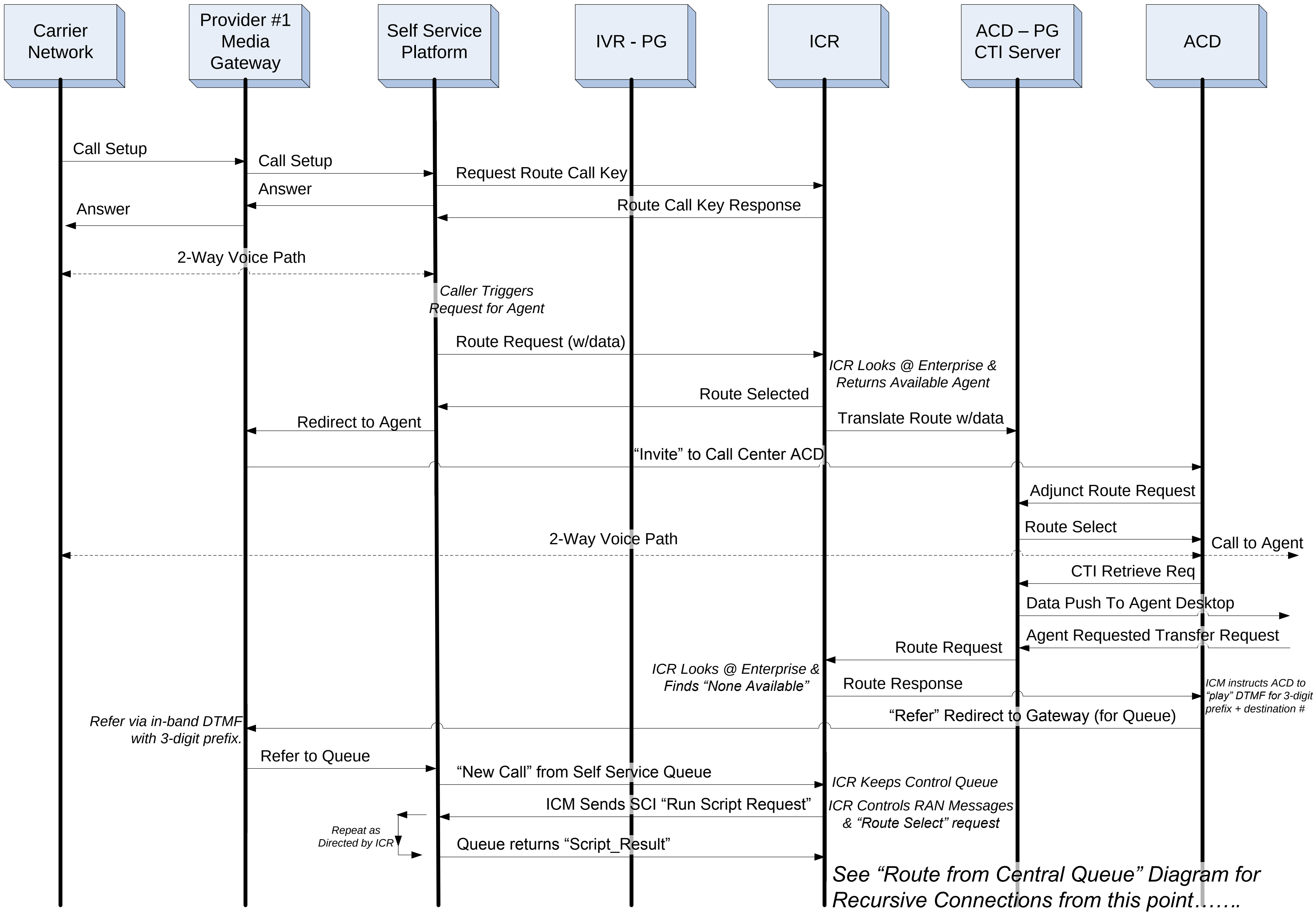
Route Call From Central Queue



Blind Transfer



Blind Transfer with Central Queue



Appendix B – Response to RIO Details Call Flow

The "RIO Requirements" provided detailed steps associated with call flow which Provider #1 has reviewed with respect to possible changes in a Voice over IP implementation. As to the steps described we found minimal change to processing as is noted in the steps below. We have identified some likely places for differences in how some of the described steps might work as CUSTOMER X makes changes to support enhancements to call routing. These possible changes will be provided as part of any detailed solution design process.

The detailed steps of the call flow involved with this integration

1) A caller dials an enterprise toll-free number to make an account inquiry. CARRIER#1 routes call to IVR.
   - Provider #1 Response: no change

2) The network terminates the call to the IVR and provides ANI & DNIS information. The IVR takes the caller through a voice response sequence, collecting information such as the nature of the inquiry and/or the customer's account number. The IVR attempts an account lookup in the Mini Subscriber Database (MSD) based on the ANI provided by the CARRIER#1 Network, if no account is found the IVR prompts customer for Service Phone Number and/or DTV Account Number.
   - Provider #1 Response: The call is actually terminated at the Sonus Media Gateway from the carrier interface perspective. The trunk level interface to the IVR is operated per Provider #1 "internal" specifications.

3) The IVR sends the IVR PG a route request to obtain the ICM Call Router Key for the call. The IVR PG passes this request to the ICR. The request to the ICR invokes a user-defined script designed to create unique numeric identifier for the phone call. The ICM Call Router Key is created by appending the Call Router Key and Call Router Key Date together.
   - Provider #1 Response: no change

4) The ICR provides a response containing the ICM Call Router Key back to the IVR via the IVR PG.
   - Provider #1 Response: no change

5) If the caller requests to speak with an agent, the IVR sends the IVR PG a route requests. The request to the ICR invokes a user-defined routing script to select the most appropriate ACD. Concurrent with the call route request sent to the IVR, the IVR sends call data to the IVR PG. The call data includes but is not limited to ANI, DNIS, Account Number, Service Phone Number, ICM Call Router Key, IVR Module, IVR Application, and IVR Reason Code.
   - Provider #1 Response: no change

6) IVR sends the sends route request to obtain wait-time information. The request invokes a user defined script designed to provide the estimated wait-time for the caller. The ICM obtains the value in seconds for the maximum delay for the requested call type.
   - Provider #1 Response: no change

7) ICR returns a response containing the estimated wait-time value back to the IVR via the IVR PG. If the wait-time exceeds 30 seconds the IVR plays an approximate wait-time value to the caller prior to transferring the call.
   - Provider #1 Response: no change

8) The ICR identifies the appropriate ACD (call center/Skill group) based on business rules such call type, minimum expected delay (MED), and longest agent available (LAA) and returns a route response label to the IVR. This is a translation route used to link the account data with routed call when it arrives at the ACD and sends a response back to IVR which contains the a label or destination via the IVR PG.
   - Provider #1 Response: no change

9) The IVR requests an CARRIER#1 Transfer Connect by playing DTMF tones including the *8 and ten-digit toll free number (skip to 11).
   - Provider #1 Response: The IVR does not actually play the tones. As described in earlier sections of this response, the IVR makes the request for transfer to the Media Gateway that looks at the number destination and use the appropriate transfer mechanism for TDM or IP destinations.

10) If the IVR PG is unavailable IVR transfers the call to a default toll-free number for the matching call type using the Digital Dashboard. The network sends a route request to the ICM software through the network interface controller (NIC). The request includes the dialed number (DN) and the calling line ID (CLID). The ICR identifies the appropriate ACD (call center/Skill group) based on business rules such call type, minimum expected delay (MED), and longest agent available (LAA) and returns a route response label to the IVR. The ICM instructs CARRIER#1 network to deliver the call to a specific ACD. (skip to ??)
   - Provider #1 Response: no change

11) CARRIER#1 invokes (completes) the transfer and sends the call to the ACD Queue (Skill Group) at the requested destination using the toll-free number provided by the ICR.
   - Provider #1 Response: no change
   - If IP, there is no Transfer Connect.

12-16 are messaging tasks getting the call to the correct agent. No Provider #1 involvement after providing Route Request with data and Route Call Key.

12) ICR notifies the ACD PG about the call being offered and also passes the call context information.
   - Current Provider #1 Involvement: None

13) The ACD routes the call to a specific Mainbank EC.
   - Current Provider #1 Involvement: None

14) The Avaya S8700 notifies CTI OS via the PG about the call being offered to a RIO CTI EC.
    - Current Provider #1 Involvement: None

15) The CTI OS Server sends the call context data to the Siebel desktop application through the Cisco CTI Driver for Siebel 7 and the Siebel Communications Server infrastructure. (The screen pop can be configured to occur either when the agent phone rings or when the agent answers).
    - Current Provider #1 Involvement: None

16) The agent answers the call using the Siebel client on their desktop. Agents execute all call control functions (such as answer, hang up, hold, transfer, etc.) directly through the Siebel browser client using the integrated multi-channel Siebel communications toolbar.
    - Current Provider #1 Involvement: None

17) The Mainbank EC initiates a Blind Transfer to Technical Tier 1 from the RIO application pop-up transfer applet. The EC's RIO desktop application checks the DEF file CTI configuration in order to identify the appropriate device command. The RIO application sends the TransferMute CTI command along with the VDN number as a parameter to the CommSessionMgr and Cisco Driver.
    - Current Provider #1 Involvement: None

18) The CUSTOMER X Siebel CTI Integration BS creates a transfer Service Request.
    - Current Provider #1 Involvement: None

19) The RIO application launches the "SetCallVariablesFromCTI" script associated with the Blind Transfer command in the DEF file CTI configuration. The script updates the ICM Call Variable as following; CallVariable1=Account Number Row Id, CallVariable2=CommEvent Id Row Id, CallVariable3=DNIS, and CallVariable4=Destination Skill Group,. The other ICM call variables will maintain the values populated on the original inbound call such as; CV5=ICM Call Router Key, CV6= IVR transfer indicator, CV7=IVR Application, CV8=IVR Module, CV9=IVR Reason Code, CV10=Siebel Bookmark.
    - Current Provider #1 Involvement: None

20) The Cisco Driver passes the Siebel Call Context information including views, screens, and open forms to Cisco Data Store (CDS) running on Siebel Gateway Server. The CDS passes a Siebel bookmark back to the Cisco Driver. The Cisco Driver passes the all call related data along with the CTI command to the CTI OS server process.
    - Current Provider #1 Involvement: None

21) The CTI OS passes the command and parameter onto the PG.
    - Current Provider #1 Involvement: None

22) The PG instructs ACD to transfer the call from the EC's telephone to a VDN on the local ACD
    - Current Provider #1 Involvement: None

23) PBX/ACD transfers call from EC's phone to VDN (Route Point) on PBX. PBX sends route request to PG

- Current Provider #1 Involvement: None

24) PG requests route instructions from the ICR, and sends it the call context information. This call context data includes ANI, DNIS and Call Variables some of which were populated by the IVR and some of which were populated by the Siebel Desktop.

- Current Provider #1 Involvement: None

25) The ICR first checks to see if an EC is available with the required skill at the local call center. If an EC is available the ICR returns a response containing a label that instructs the PBX to send the call to the appropriate ACD Queue (Skill Group) at the local call center.

- Current Provider #1 Involvement: None

26) If an EC is not available for the required skill at the local call center, the ICR first checks the value in CV6 in order to ascertain the correct call type or skill group for the call. The ICR then identifies the appropriate ACD (call center/Skill group) based on business rules such call type, minimum expected delay (MED), and longest agent available (LAA) and returns a route response label to the local ACD. For this scenario the ICR target Call Center #2.

- Current Provider #1 Involvement: None

27) ICR sends a response back to ACD which contains a label or destination via the ACD PG. This is a translation route used to link the account data with routed call when it arrives at the remote ACD. The label contains the dialing instructions, for initiating a *8 transfer and the toll-free number associated with the temporary DNIS at the destination call center

- Current Provider #1 Involvement: None

28) The PG instructs the ACD to dial the *8 and the toll-free number.

- Current Provider #1 Involvement: None

29) ICR sends final target and call context to receiving PG at the remote call center.

- Current Provider #1 Involvement: None

30) The Avaya S8700 PBX/ACD plays the appropriate DTMF and initiates a Transfer Connect in CARRIER#1 Network. CARRIER#1 invokes the transfer and sends the call to the temporary DNIS on the ACD at the requested destination using the toll-free number provided by the ICR.

- Current Provider #1 Involvement: None

31) When the call arrives to the temporary target, the Avaya S8700 makes an adjunct route-request to the PG to link the call data captured earlier and to receive the target skill group for the call. The receiving PG recognizes the DN, and returns the final target DNIS

- Current Provider #1 Involvement: None

32) The remote call center ACD routes the call to a specific Technical Tier 1 EC. The Avaya S8700 notifies PG about the call being offered to a desktop agent. The PG alerts the Siebel

desktop of the offered call through CTI OS and the Cisco CTI Driver for Siebel 7. The Cisco Driver retrieves the call context information from the CDS using the bookmark

- Current Provider #1 Involvement: None

33) The CTI OS Server sends the call context data to the Siebel desktop application through the Cisco CTI Driver for Siebel 7 and the Siebel Communications Server infrastructure. (The screen pop can be configured to occur either when the agent phone rings or when the agent answers).

- Current Provider #1 Involvement: None

34) The agent answers the call using the Siebel client on their desktop. Agents execute all call control functions (such as answer, hang up, hold, transfer, etc.) directly through the Siebel browser client using the integrated multi-channel Siebel communications toolbar.

- Current Provider #1 Involvement: None