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ACE Direct Platform Release Documentation

ACE Direct Installation and Configuration Guide

Version 6.0

July 23, 2021

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Record of Changes

| Version | Date | Author/Owner | Description of Change |
| --- | --- | --- | --- |
| 1.0 | November 4, 2016 | CMS Alliance to Modernize Healthcare | Version 1.0 for release to Sponsor |
| 1.1 | February 17, 2017 | CMS Alliance to Modernize Healthcare | Version 1.1 for release to Sponsor |
| 2.0 | November 1, 2017 | CMS Alliance to Modernize Healthcare | Version 2.0 for release to Sponsor |
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1. Executive Summary

The Federal Communications Commission (FCC) Telecommunications Relay Service (TRS) Center of Expertise (COE) Project promotes the Commission’s goal to foster innovations that advance functionally equivalent telecommunications. Toward that end, the project ensures that the Telecommunications Relay Service employs improved technology for persons who are d/Deaf, Hard of Hearing, DeafBlind, and/or have speech disabilities. The FCC has embraced a research-based approach to achieve this goal by engaging the Centers for Medicare & Medicaid Services (CMS) Alliance to Modernize Healthcare Federally Funded Research and Development Center (the Health FFRDC), operated by The MITRE Corporation (MITRE), to conduct independent engineering assessments that promote and demonstrate TRS’s functional equivalence.

The Health FFRDC is independently assessing voice telephone services, video access services, and Internet Protocol (IP)-based captioning technology; improvements to TRS efficiency; solutions for direct communication between people with communication disabilities and other telephone users; and the effectiveness, efficiency, and consumer response to current and future approaches for delivering TRS.

At the FCC’s request, the Health FFRDC developed a Direct Video Calling (DVC) Auto-Routing Proof of Concept (POC) in support of the FCC’s Accessible Communications for Everyone (ACE)[[1]](#footnote-2) program. This DVC auto-routing platform enables direct calling from d/Deaf or hard of hearing individuals to an American Sign Language (ASL)-trained agent within the organization’s call center. The agent handles the call using a video-capable phone with real-time video connection. To demonstrate the capabilities of DVC, the FCC and the Health FFRDC have further advanced the original auto-routing POC into a call center platform for 2 to 20 customer service representatives. This new DVC platform is called ACE Direct.

Table ES-1 describes the new features released in this version of ACE Direct. Subsection 2.4 provides a complete history of ACE Direct releases and their associated features.

Table ES-1. New ACE Direct Features

| Version | Release Date | New Infrastructure Feature or Capability |
| --- | --- | --- |
| 6.0 | July 23, 2021 | * **Call Monitoring** – The Agent portal now allows for an Agent to passively monitor an ongoing call between another Agent and a consumer. * **Call Transfers** – The Agent portal now supports the transferring of calls between Agents. * **Call Recording** – The Agent portal now supports recording of a call. Call recordings can be used for training purposes or to record behavior that may concern the agent. * **File-Sharing Enhancements** – ACE Direct now supports a more secure file sharing between the Agent and Consumer. An open-source virus and malware scanner (ClamAV) has been integrated into the ACE Direct file sharing function. * **Multi-Party Captions and Chat** – The Agent and Consumer portals will display captions and chat in multi-party calls. * **WebRTC In-Call Statistics and FPS Indicator** – The agent portal can retrieve and log WebRTC in-call statistics as a diagnostics tool. New frames per second and packets lost indicators appear on the agent portal during a call. |
| 5.0 | February 5, 2021 | * **No changes to the installation procedures were released** |
| 4.0 | July 31, 2019 | * **Kurento Media Server** –The media server provides advanced media processing, recording, monitoring and bandwidth controls. |
| 3.1 | April 9, 2019 | * **SIP Proxy Server** – The SIP Proxy server provides a single point of entry following Defense-in-Depth principles to create a layer between the ACE Direct environment and the Internet. This enhanced security provides a means to mitigate certain exploits and Distributed Denial of Service (DDoS) attacks. |
| 3.0 | October 26, 2018 | * **Containers –** Containers simplify the overall ACE Direct installation, configuration, and deployment. They improve portability to different environments and add modularity. * **Management Portal Agent Provisioning UI** – The Management Portal Agent Provisioning screen makes it easy for call center managers to provision and maintain agent users in both Open Access Management (OpenAM) and ACE Direct. This allows customization of the default agent accounts. * **Data Logger Utility** – The Data Logger Utility captures and saves log files, trace information, and testing details automatically. This information facilitates troubleshooting interoperability and call quality issues. * **NGINX Custom Error Page** – The NGINX Custom Error Page is a more user-friendly page than the default NGINX error page. This ACE Direct page appears when the system is offline. * **ASL Video On Hold** – This feature allows the call center to display or advertise a custom message to a caller while on hold or after hours. * **Customizable ACE Direct URLs** – Customizable ACE Direct URLs allow owners, like the FCC, to customize the public URLs to match their corporate name or brand. An example is https://xyzcorp.org/XYZDirect/agent. |

Implementing the Direct Video Calling platform provides critical benefits toward achieving functionally equivalent telecommunications:

* **Improved Communications** – DVC improves privacy and decreases misrepresentation, which improves efficiency, effectiveness, and productivity.
* **Career Opportunities** – Employing native ASL users to handle customer service video calls expands hiring opportunities. Executive Order 13548 (July 2010) directed federal agencies to increase employment opportunities for people with disabilities.
* **Simple Implementation** – The technology to implement a DVC system is readily obtainable, affordable, and easy to set up.
* **Secure Communications** – With proper configuration, agencies can use high-speed broadband and their own internal networks without compromising security or contending with barriers created by firewalls.
* **Maintain ADA Compliance** – DVC ensures compliance with the Americans with Disabilities Act mandates.
* **Cost Savings** – Replacing three-way interpreted calls with two-way direct communication saves money by minimizing the need for repeat calls due to miscommunication and/or misunderstanding.

As part of this effort, the Health FFRDC developed and documented requirements and features, including user stories and associated use cases. The Health FFRDC also configured, tested, and integrated provider endpoint video devices with the ACE Direct platform. Detailed configuration and source code files are available for download and reproduction to improve solutions to support the community. The public can download or clone these files at <https://github.com/FCC/ACEDirect>.

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# Introduction

The Federal Communications Commission (FCC) Telecommunications Relay Service (TRS) Center of Expertise (COE) Project promotes the Commission’s goal to foster innovations that advance functionally equivalent telecommunications. Toward that end, the project ensures that the Telecommunications Relay Service employs improved technology for persons who are d/Deaf, Hard of Hearing, DeafBlind, and/or have speech disabilities. In this document, “d/Deaf” describes individuals who are deaf in the audiological sense as well as those who identify as culturally Deaf.

The CMS Alliance to Modernize Healthcare Federally Funded Research and Development Center (the Health FFRDC), sponsored by the Centers for Medicare & Medicaid Services (CMS) and all divisions of the Department of Health and Human Services (HHS), is the first FFRDC dedicated to strengthening the nation’s healthcare system. The MITRE Corporation (MITRE), an objective not-for-profit organization, operates the Health FFRDC in partnership with CMS and all HHS agencies to implement innovative ideas to solve our nation’s toughest health problems.

## Background

The FCC has embraced a research-based approach to achieve this goal by engaging the Health FFRDC to conduct independent engineering assessments that promote and demonstrate TRS’s functional equivalence. As part of the Accessible Communications for Everyone (ACE) program, the Health FFRDC independently assesses voice telephone services, video access services, and Internet Protocol (IP)-based captioning technology; improvements to TRS efficiency; solutions for direct communication between people with communication disabilities and other telephone users; and the effectiveness, efficiency, and consumer response to current and future approaches for delivering TRS.

In continuing pursuit of the Commission’s goal to advance functionally equivalent telecommunications, the Health FFRDC developed ACE Direct, an open-source call center platform that supports Direct Video Calling (DVC) for 2 to 20 Agents. Implementing ACE Direct in a corporate production environment requires customization to ensure adherence to corporate practices and policies related to security, system configurations, cloud services, and availability.

The FCC encourages government agencies and private businesses to make DVC part of their call center strategy because it offers significant gains for providing functionally equivalent telecommunications, including:

* **Improved Communications** – DVC improves privacy and decreases misrepresentation, which enhances efficiency, effectiveness, and productivity.
* **Career Opportunities** – Employing native American Sign Language (ASL) consumers to handle customer service video calls expands hiring opportunities. Executive Order 13548 (July 2010) directed federal agencies to increase employment opportunities for people with disabilities.
* **Simple Implementation** – The technology to implement a DVC system is readily obtainable, affordable, and easy to set up.
* **Secure Communications** – With proper configuration, agencies can use high-speed broadband and their own internal networks without compromising security or contending with barriers created by firewalls.
* **Maintain ADA Compliance** – DVC ensures compliance with the Americans with Disabilities Act (ADA) mandates.
* **Cost Savings** – Replacing three-way interpreted calls with two-way direct communication saves money by minimizing the need for repeat calls due to miscommunication and/or misunderstanding.

The Health FFRDC developed and documented ACE Direct requirements and features, including consumer stories and associated use cases. The FFRDC also configured, tested, and integrated provider endpoint video devices using the ACE Direct platform.

## Purpose and Scope

This document is an installation and configuration guide for the components that make up the ACE Direct system. It describes how to install, configure, and deploy the entire ACE Direct application.

In addition to this release documentation, detailed configuration and source code files are available to the public at <https://github.com/FCC> for download and reproduction of the platform to support and promote future platform enhancements for the d/Deaf, Hard of Hearing, DeafBlind, and speech-disabled community.

# Installation and Configuration

This section presents guidance for the installation and configuration of the ACE Direct components except for Open Access Management (OpenAM) and the Kuando Busylight™. To reproduce this version of the platform, review the README.md file in the autoinstall folder at <https://github.com/FCC>.

Table 1 presents a list of initial requirements that must be met before installation.

Table 1. Initial Installation Requirements for ACE Direct

| Prerequisite | Rationale |
| --- | --- |
| A domain name (your domain name must not contain an underscore) | Needed to provide end users with access to the ACE Direct Platform via a web browser. |
| Second-level Domain Name System (DNS) subdomain names and corresponding Address Records (A records) (e.g., “host.example.com”) | Needed to point the subdomains to a specific IP address used in components of ACE Direct. |
| Servers must have external Internet access | Needed for external connectivity and public-facing components. |
| The NGINX/ Session Traversal Utilities for NAT (STUN)/ Traversal Using Relay NAT (TURN) servers have a public and local IP address | Needed to facilitate application network traffic. |
| SELinux and IPv6 disabled on all hosts | ACE Direct does not currently support these technologies. |
| A “dial-in” number that has been registered in Internet Telecommunications Relay Service (iTRS) and/or a Session Initiation Protocol (SIP) trunk provider (such as Twilio) | To allow Consumers to place calls to ACE Direct. **Note:** FCC rules govern access to the iTRS database. |
| IPtables disabled for Asterisk and NGINX | Disable if not required. |
| Wildcard certificate (preferred) | Allows a single certificate to be used for all subdomains; otherwise, each Fully Qualified Domain Name (FQDN) will need its own certificate. |
| Create ‘acedirect’ user on servers | Required for node.js installation. |
| Access to system package mirrors (Needed for ‘yum install’) | Needed to install the necessary package dependencies for the ACE Direct Platform. |
| Chrome browser is required for Agent Desktop | Google Chrome browser leads the industry in Web Real-Time Communication (WebRTC) integration. Additionally, Chrome is used for all ACE Direct testing and provides the high level of stability. |
| Purchase Busylight™ hardware (optional) | The Busylight™ provides call centers with a visual indication of the Agent’s status. |
| Google Cloud Platform Account or IBM Cloud Account (both optional) | Required to support optional captioning and language translation features. |

## Secure Sockets Layer/Transport Layer Security Certificate

The applications within ACE Direct use Hypertext Transport Protocol Secure (HTTPS) to provide security during web transmissions. To prevent major web browsers from potentially flagging the application as untrusted, it is recommended to acquire and implement a Secure Sockets Layer/Transport Layer Security (SSL/TLS) certificate from a trusted certificate authority (CA), such as LetsEncrypt (<https://letsencrypt.org/>). After obtaining a certificate, follow the instructions in the respective README.md files in the various ACE Direct repositories to install the SSL certificate. ACE Direct also employs the [HTTP](https://en.wikipedia.org/wiki/HTTP) Strict Transport Security (HSTS) web policy mechanism to protect against [protocol downgrade attacks](https://en.wikipedia.org/wiki/Protocol_downgrade_attack) and [cookie hijacking](https://en.wikipedia.org/wiki/Session_hijacking).

## Asterisk Installation and Configuration Script

An automated installation script has been developed to assist in the deployment and configuration of the Asterisk server. The Asterisk install script will install and configure Asterisk for ACE Direct on a CentOS 7 and Amazon Linux 2 servers. This script, which is available at <https://github.com/FCC> in the ‘scripts’ directory of the Asterisk repository, can be used to quickly create an Asterisk instance. The script will perform the following tasks automatically:

* Update current packages and install required packages for Asterisk
* Install PJSIP
* Install Asterisk
* Configure and apply Asterisk configs, media, scripts, and custom patches
* Start Asterisk

It is necessary to satisfy several prerequisites to the script before installing Asterisk. These prerequisites are listed in the top-level README.md file of the Asterisk repo. Review the README.md file in the Asterisk repository before running the script.

### How It Works

The Asterisk Private Branch Exchange (PBX) system relies on the configuration of three key files in the /etc/asterisk directory: pjsip.conf, extensions.conf, and http.conf. The pjsip.conf file contains the connection endpoints and parameters, and the extensions.conf file contains the syntax and programming for the extensions assigned to those endpoint connections. The http.conf file is the server configuration for Asterisk and defines the certificates Asterisk uses for secure communications.

#### Dial Plan Configuration

The dial plan is defined by the extensions.conf and pjsip.conf files in the /etc/asterisk directory, which work together to establish the connection between devices and route the calls to those devices. An endpoint device is any device that places or receives a call. Table 2 shows an example of common endpoint devices, the associated extension, and the required codecs for the sample dial plan configuration in this guide.

Table 2. Example Asterisk Endpoint Extensions

| Extension | Purpose | Device | Codec |
| --- | --- | --- | --- |
| 30001 | ACE Direct Agent 1 | softphone | h264/vp8/ulaw |
| 30002 | ACE Direct agent 2 | softphone | h264/vp8/ulaw |
| 30003 | ACE Direct agent 3 | softphone | h264/vp8/ulaw |
| 30004 | ACE Direct agent 4 | softphone | h264/vp8/ulaw |

As displayed in Table 2, the extension is the number assigned to the endpoint. The ability of that endpoint to connect to the Asterisk instance is configured in the pjsip.conf file and the ability to dial to another endpoint is configured within the extensions.conf file. The device can be any phone capable of connecting to the Asterisk instance—a softphone, a desktop phone, a video communication phone, or other such device. At present, the configuration provided with ACE Direct can only be used with Web Real-Time Communication (WebRTC) and Video Relay Service (VRS) provider devices; other types of devices are not currently supported. The codec refers to the preferred media codec, or in some cases, the only available codec, used by the endpoint. The codec is also configured in the pjsip.conf file.

#### The extensions.conf Configuration File

The extensions.conf file is the configuration file that defines how the endpoint extensions communicate with each other or with the Asterisk server itself. The Asterisk server consists of many different applications working together to perform different functions. These functions can be called in the extensions.conf dial plan to route an endpoint device to a desired outcome. For example, a Consumer wants to dial voicemail. In extensions.conf, that configuration must state that the Consumer’s phone number is to load the voicemail application, which is done by the following syntax:

exten => \_6XXX,1,Answer()

same => n,Voicemail(1234@ourpbx)

In this example, any extension matching the pattern \_6XXX will be answered for immediate placement into the videomail application.

For specific phone numbers to route, as well as number schemes for unknown numbers, use the wildcard “X.” In the following example as shown in the context “[from-internal]” of the extensions.conf file, all extensions ended with 6xxx are configured. The “[from-internal]” label refers to the context in which those extensions reside. A context is an organizational container that can host a group of extensions and extension patterns that can route and dial to other extensions within the same context.

[from-internal]

exten => \_50XX,1,Dial(PJSIP/${EXTEN})

same => n,DumpChan()

same => n,HangUp()

exten => \_6XXX,1,Dial(PJSIP/${EXTEN})

same => n,DumpChan()

same => n,HangUp()

exten => \_70XX,1,Dial(PJSIP/${EXTEN})

same => n,DumpChan()

same => n,HangUp()

Extension patterns use wildcards to encompass a greater range of possible matches to the same string. To dial to an extension in another context, that context must be specifically identified in the coding. The context defined must also be the same for the “Context” attribute of the endpoints defined in the pjsip.conf file (please refer to Subsection 3.2.1.3). For the Dial() function call, the extensions/numbers defined within the function call must be defined in pjsip.conf, as shown in the following subsection.

#### The pjsip.conf Configuration File

The pjsip.conf file defines the endpoint connection parameters. Some of the key configuration settings in the pjsip.conf file include the transport protocol, user extensions and external IP address. In addition, the file must define the authentication method for this extension. PJSIP does not require the specification of a protocol and will dynamically select the best available option to use. The extension profile must specify the configuration settings for the endpoint, the network connection, the allowed codecs, and network address translation (NAT) traversal settings. The extension profile may also require that specific settings be disabled or not used. Visit the Digium Asterisk website to thoroughly understand the function of the pjsip.conf file and its associated settings.

The Asterisk PBX uses the pjsip.conf file to load the connection points from which the endpoint devices will acquire their information and connect. This means that this file is responsible for the endpoint connection, handling the contact that connects to that endpoint, routing the call, and passing information. The following critical attributes are defined in a profile within pjsip.conf:

* **Transport.** The transport defined within the pjsip.conf file specifies the configuration for Transmission Control Protocol (TCP), User Datagram Protocol (UDP), WebSocket (WS), or WebSocket Secure (WSS) connections. These transport settings require configuration as to the external IP, internal IP, and in the case of web sockets, certificate information to be used for secure communication.
* **Endpoint.** An endpoint in the pjsip.conf file is a profile that matches to an endpoint device. The endpoint can be defined in pjsip.conf independently or as a template that numerous extensions can call. This template helps to maintain a condensed file, otherwise it can grow quite large. Endpoint templates are identified by (!) after the identifying name.
* **Register.** A registration is a type of authentication from an endpoint device. The device will send authentication information to Asterisk that will then “register” the device to the PBX with the assigned endpoint profile. A registration is a temporary assignment of configuration options and network settings to identify the endpoint device and route calls.
* **Contact.** A contact is the network-identifying information of a device that has registered and its corresponding extension. Viewing the contact information within the Asterisk console verifies that an endpoint device has authenticated and been assigned its configuration correctly.

Once the profile has been defined, extensions and numbers can be associated with the profile, which will be referenced in the extensions.conf file to help route calls to their proper destination. Use the pre-defined profiles in the Asterisk repository because they are already configured specifically for use with ACE Direct.

### Web Secure Sockets

This Consumer portal uses WebRTC technology to conduct video, voice, and audio communications. Asterisk must use web sockets and for most browsers, secure web sockets to successfully use WebRTC technology. Asterisk should only use a certificate from a valid certificate authority when using WebRTC.

### Sample Configuration Files

**Note:** The following samples are not up to date with the latest configuration in the Asterisk repository. The intent is to provide a basic overview of some of the configuration and their associated parameters. To view the latest configuration files, visit the Asterisk GitHub repository.

#### Sample pjsip.conf Configuration

The following is an example of the pjsip.conf file used by Asterisk:

---------------------------------top of pjsip.conf file--------------------------------------------------

[transport- wss] ;name of transport configuration

type=transport ; type being configured is of type transport

protocol= wss ;protocol is web secure socket, can be tcp,udp,ws,wss

bind=0.0.0.0:443 ;bound ip/port

external\_media\_address=<IP ADDRESS> ;external ip of Asterisk

cert\_file =/etc/asterisk/keys/star.pem ;certificate pem

priv\_key\_file=/etc/asterisk/keys/star.key ;certificate key

[endpoint-basic](!) ;Defines the name of the endpoint, the (!) designates it as a template

type= endpoint ;defines the type

transport=transport-wss ;defines the transport to be used

context= from-internal ;defines the context, must correspond to context in extensions.conf

disallow= all ;explicitly deny all media unless specified

allow= ulaw ;specify allow ulaw audio codec

allow=vp8 ;specify allow vp8 video

allow=h264 ;specify h264 video

allow=t140 ;specify allow t140 text protocol

force\_rport=yes ;network configuration, reflexive port

direct\_media=no ;network configuration, do not establish direct media link (NAT)

rewrite\_contact=yes

media\_address=<IP ADDRESS>

rtp\_symmetric=yes

ice\_support=yes

message\_context =internal-im ;context for internal messenger

[30001](endpoint-basic) ;Extension information, extension username is 30001

auth =auth30001 ;The auth profile to use

aors =30001 ;The aors profile to use

[auth30001](auth-userpass)

password= changeit! ;password to be changed

username=30001 ;username for authentication, typically extension number

[30001](aor-single-reg)

remove\_existing =yes ; Remove pre-existing contact

max\_contacts =1 ;Number of simultaneous contacts registered to an AOR

qualify\_frequency =5 ;Interval in seconds of qualifying a registered contact

authenticate\_qualify=yes ;Send authentication request on qualify if required

---------------end of pjsip.conf file----------------------------

#### WebRTC Sample Configuration

The following is a sample WebRTC-enabled endpoint template in pjsip.conf:

[endpoint-webrtc](!)

type=endpoint

transport=transport-wss

context=from-internal

disallow=all

allow=ulaw

allow=h264

allow=vp8

allow=t140

force\_avp=yes

use\_avpf=yes

media\_encryption=dtls

dtls\_verify=fingerprint

dtls\_fingerprint=SHA-1

dtmf\_mode=auto

dtls\_rekey=0

dtls\_cert\_file=/etc/asterisk/keys/asterisk.pem

dtls\_ca\_file=/etc/asterisk/keys/ca.crt

dtls\_setup=actpass

ice\_support=yes

media\_use\_received\_transport=yes

rtp\_symmetric=yes

force\_rport=yes

rewrite\_contact=yes

message\_context=internal-im

rtcp\_mux=yes

## Node.js

Node.js is a platform built on Chrome’s V8 JavaScript run-time engine, which was developed to build fast and scalable network applications. Node.js is event driven, lightweight, and efficient—ideal for data-intensive real-time applications that run across distributed devices.

Node.js servers provide the functionality of ACE Direct. More specifically, the servers host the Agent, Consumer, Management, and Videomail portals. Node.js also provides Agent Data and Provider Data through RESTful application programming interfaces (API) to ACE Direct.

Examples of these RESTful services are VRS lookup and Agent verify functions. An Asterisk instance should already be up and running to support most of the Node.js instances.

Instructions for downloading and installing Node.js can be found on the official Node.js website (<https://nodejs.org/en/>). For this version of ACE Direct, the Node.js server is built with the following software versions:

* Operating System: CentOS 7.x
* Node.js: Version 12.18.2

### Node.js Components Installation

The manual installation procedures for the Node.js components of ACE Direct have been deprecated and are no longer supported. To install the Node.js components, follow the README.md file in the autoinstall folder located at <https://github.com/FCC>.

## Management Portal

The ACE Direct Management Portal provides key performance indicators for real-time monitoring by the call center Manager. This information may be used to improve user experience, increase user satisfaction, and support monitoring of overall call center performance. Some of the configuration definitions can be found in parameter\_desc.json of the dat folder and in the queues.conf in the asterisk repository folder.

### Log Files

Log files are in managementportal/logs. The debug level is defined by the debug\_level field in the configuration file. The valid options are as follows: ALL, TRACE, DEBUG, INFO, WARN, ERROR, and FATAL.

### Management Dashboard

#### Web Server and Client Configuration

The Management Dashboard functionality relies on the Asterisk server to collect reports on call agents and queue status. The Management Dashboard (dashboard.js and dashboard.ejs) is hosted on the Node.js server (server-db.js). The following subsections describe the operational data flow.

##### Server Side

* **server-db.js** – Communicates with the Asterisk Server through the Asterisk Management Interface (AMI) protocol. AMI events provide the Management Portal with data on queue and agent status.
* Client Side
* **dashboard.ejs** – Controls the User Interface (UI) for the Management Portal Dashboard.
* **dashboard.js** – Uses JavaScript data structures to store information to be displayed on the dashboard.ejs page.

#### Asterisk Management Interface

The AMI allows a client application to connect to an Asterisk instance and issue actions (requests) and receive events (responses). The Management Portal performs eight unique AMI action calls to collect data on queue and agent status. Table 3 shows the AMI actions and descriptions.

Table 3. AMI Actions and Descriptions

| AMI Action | Description | Syntax | Arguments | Triggered Events |
| --- | --- | --- | --- | --- |
| Agents | Queries for information about all agents | Action: Agents ActionID: <value> | * ActionID – ActionID for this transaction. Will be returned. | Agents  AgentsComplete |
| DBDel | Deletes an entry from the Asterisk internal database | Action: DBDel ActionID: <value> Family: <value> Key: <value> | * ActionID – ActionID for this transaction. Will be returned. * Family - Name of variable * Key - Value | Call block |
| DBGet | Retrieves an entry from the Asterisk internal database | Action: DBGet ActionID: <value> Family: <value> Key: <value> | * ActionID – ActionID for this transaction. Will be returned. * Family - Name of variable * Key - Value | Call block |
| DBPut | Adds an entry to the Asterisk internal database | Action: DBPut ActionID: <value> Family: <value> Key: <value> Val: <value> | * ActionID – ActionID for this transaction. Will be returned. * Family - Name of variable * Key - Value * Val - Boolean | Call block |
| Queues | Queries for queues information | Action: Queues |  | Queues |
| QueueReset | Resets the running statistics for a queue | Action: QueueReset ActionID: <value> Queue: <value> | * ActionID – ActionID for this transaction. Will be returned.   Queue – The name of the queue on which to reset statistics. |  |
| QueueStatus | Check the status of one or more queues | Action: QueueStatus ActionID: <value> Queue: <value> Member: <value> | * ActionID – ActionID for this transaction. Will be returned. * Queue – Limit the response to the status of the specified queue. * Member – Limit the response to the status of the specified member. | QueueStatus  QueueStatusComplete  QueueMember  QueueParams |
| QueueSummary | Requests Asterisk to send a QueueSummary event | Action: QueueSummary ActionID: <value> Queue: <value> | * ActionID – ActionID for this transaction. Will be returned. * Queue – Queue for which the summary is requested. | QueueSummary  QueueSummaryComplete |

The dashboard server listens for the Asterisk AMI events shown in Table 4.

Table 4. Asterisk AMI Events and Descriptions

| AMI Action | Description | Syntax | Fields |
| --- | --- | --- | --- |
| AgentsComplete | Generated when an agent has finishes servicing a member in the queue | Event: AgentComplete Queue: <value> Member: <value> MemberName: <value> HoldTime: <value> [Variable:] <value> TalkTime: <value> Reason: <value> Queue: <value> Uniqueid: <value> Channel: <value> Member: <value> MemberName: <value> HoldTime: <value> | * Queue – The name of the queue. * Member – The queue member's channel technology or location. * MemberName – The name of the queue member. * HoldTime – The time the channel was in the queue, expressed in seconds since 00:00, Jan 1, 1970 UTC. * Variable – Optional channel variables from the ChannelCalling channel * TalkTime – The time the agent talked with the member in the queue, expressed in seconds since 00:00, Jan 1, 1970 UTC. * Reason * consumer * agent * transfer * Queue * Uniqueid * Channel * Member * MemberName * HoldTime |
| AgentLogin | Generated when an agent logs in | Event: AgentLogin Agent: <value> Channel: <value> Uniqueid: <value> | * Agent – The name of the agent. * Channel * Uniqueid |
| AgentLogoff | Generated when an agent logs off | Event: AgentLogoff Agent: <value> Agent: <value> Logintime: <value> Uniqueid: <value> | * Agent – The name of the agent. * Agent * Logintime * Uniqueid |
| FullyBooted | Generated when all Asterisk initialization procedures complete | Event: FullyBooted Status: <value> | * Status |
| QueueMemberAdded | Generated when a new member is added to the queue | Event: QueueMemberAdded Queue: <value> Location: <value> MemberName: <value> StateInterface: <value> Membership: <value> Penalty: <value> CallsTaken: <value> LastCall: <value> Status: <value> Paused: <value> Queue: <value> Location: <value> MemberName: <value> StateInterface: <value> Membership: <value> Penalty: <value> CallsTaken: <value> LastCall: <value> Status: <value> Paused: <value> | * Queue – The name of the queue. * Location – The queue member's channel technology or location. * MemberName – The name of the queue member. * StateInterface – Channel technology or location from which to read device state changes. * Membership * dynamic * realtime * static * Penalty – The penalty associated with the queue member. * CallsTaken – The number of calls this queue member has serviced. * LastCall – The time this member last took call, expressed in seconds since 00:00, Jan 1, 1970 UTC. * Status – The numeric device state status of the queue member. * 0 - AST\_DEVICE\_UNKNOWN * 1 - AST\_DEVICE\_NOT\_INUSE * 2 - AST\_DEVICE\_INUSE * 3 - AST\_DEVICE\_BUSY * 4 - AST\_DEVICE\_INVALID * 5 - AST\_DEVICE\_UNAVAILABLE * 6 - AST\_DEVICE\_RINGING * 7 - AST\_DEVICE\_RINGINUSE * 8 - AST\_DEVICE\_ONHOLD * Paused * 0 * 1 * Queue * Location * MemberName * StateInterface * Membership * Penalty * CallsTaken * LastCall * Status * Paused |
| QueueMemberRemoved | Generated when a queue member is removed from the queue | Event: QueueMemberRemoved  Queue: <value> Location: <value> MemberName: <value> Queue: <value> Location: <value> MemberName: <value> | * Queue – The name of the queue. * Location – The queue member's channel technology or location. * MemberName – The name of the queue member. * Queue * Location * MemberName |

### Call Detail Record Dashboard

The Asterisk PBX system can collect and store call data for each call that traverses the system. These data records, referred to as call detail records (CDRs), are collected and stored for use by business intelligence (BI) tools, statistical analysis, and even compensation and reimbursement. The information collected is invaluable and can be retrieved and stored using various methods.

By default, the Asterisk server stores CDRs in a file called master.csv located in /var/log/asterisk/cdr-csv. CDR records can be collected in real time, or near-real time in various ways. The Asterisk server also supports different database connections. One method is to use Asterisk on an internal application for connecting to a database. Another is to configure an Open Database Connectivity (ODBC) connection for Asterisk. There is no restriction on the type of relational database used. For example, MySQL, MariaDB, and others are all viable options. The Health FFRDC elected to use a MySQL database server with Asterisk connecting through an ODBC connection.

#### MySQL Installation

To start, it is necessary to install additional packages for the MySQL server and the ODBC connector as follows:

sudo yum install unixODBC unixODBC-devel libtool-ltdl libtool-ltdl-devel mysql-connector-odbc  
wget http://repo.mysql.com/mysql-community-release-el7-5.noarch.rpm  
sudo rpm -ivh mysql-community-release-el7-5.noarch.rpm  
sudo yum install mysql-server  
sudo systemctl start mysqld  
sudo yum update

Once the packages are installed, it is necessary that Asterisk recognizes these new packages. To do so. migrate to the Asterisk directory, which in the MITRE build is in /usr/src/asterisk-16.8.0/.

From this directory, run:

./configure  
make menuselect

Once the graphic window is displayed, check to make sure that the res\_odbc module is selected. Save and Exit the graphic window.

Next, run the following:

make && make install

This will configure the Asterisk installation for use with the ODBC connection. Once the installation script finishes successfully, proceed to start the MySQL server.

Once MySQL has been installed, run the basic hardening script for MySQL to remove anonymous connections, the test database, and establish the root password. To do this, run the following script and follow the prompts:

sudo /usr/bin/mysql\_secure\_installation

#### Database Configuration

After installing MySQL and completing the initial configuration, log into MySQL using:

mysql -u root -p

Once logged in, create the database and the table needed to store the CDRs. To do so, run the following commands:

CREATE DATABASE asterisk;

USE asterisk;   
   
CREATE TABLE `bit\_cdr` (

`calldate` datetime NOT NULL default '0000-00-00 00:00:00', `clid` varchar(80) NOT NULL default '',

`src` varchar(80) NOT NULL default '',

`dst` varchar(80) NOT NULL default '',

`dcontext` varchar(80) NOT NULL default '',

`channel` varchar(80) NOT NULL default '',

`dstchannel` varchar(80) NOT NULL default '',

`lastapp` varchar(80) NOT NULL default '',

`lastdata` varchar(80) NOT NULL default '',

`duration` int(11) NOT NULL default '0',

`billsec` int(11) NOT NULL default '0',

`disposition` varchar(45) NOT NULL default '',

`amaflags` int(11) NOT NULL default '0',

`accountcode` varchar(20) NOT NULL default '',

`userfield` varchar(255) NOT NULL default '',

`uniqueid` VARCHAR(32) NOT NULL default '',

`linkedid` VARCHAR(32) NOT NULL default '',

`sequence` VARCHAR(32) NOT NULL default '',

`peeraccount` VARCHAR(32) NOT NULL default '' );   
   
ALTER TABLE `bit\_cdr` ADD INDEX ( `calldate` );   
ALTER TABLE `bit\_cdr` ADD INDEX ( `dst` );   
ALTER TABLE `bit\_cdr` ADD INDEX ( `accountcode` );

To create a user account for remote connections, use the following syntax:

CREATE USER 'username'@'%' IDENTIFIED BY 'password';

GRANT ALL PRIVILEGES ON \*.\* TO 'username'@'%' WITH GRANT OPTION;

FLUSH PRIVILEGES;

#### Asterisk Integration

Now that the database is up and running, connect to Asterisk by using the ODBC connector, as follows:

vi /etc/odbcinst.ini

[MySQL]

Description = ODBC for MySQL

Driver = /usr/lib/libmyodbc5.so

Setup = /usr/lib/libodbcmyS.so

Driver64 = /usr/lib64/libmyodbc5.so

Setup64 = /usr/lib64/libodbcmyS.so

FileUsage = 1

vi /etc/odbc.ini

[asterisk-connector]

Description = MySQL connection to 'asterisk' database

Driver = MySQL

Database = asterisk

Server = localhost

User = root

Password = password

Port = 3306

Socket = /var/lib/mysqld/mysqld.sock

To test your configuration, run the following command from the CLI:

echo "select 1" | isql -v asterisk-connector

A message of Connected! should be returned.

| Connected! |

SQLRowCount returns 1 1 rows fetched

Now you will need to point Asterisk to your new database. To do so, modify the /etc/asterisk/res\_odbc.conf file with the database name, username, password, and port of the ODBC connection you have set up. The dsn variable will point to the dsn identified:

vi /etc/asterisk/res\_odbc.conf  
  
[asterisk]  
enabled => yes  
dsn => asterisk-connectorusername => username  
password => password  
pooling => no  
limit => 99999  
pre-connect => yes

Next, edit /etc/asterisk/cdr\_odbc.conf to include the following:

vi /etc/asterisk/cdr\_odbc.conf  
  
[global]  
dsn=asterisk  
loguniqueid=yes  
table=bit\_cdr

**Note:** the dsn variable will point to the DSN identified in res\_odbc.conf, NOT /etc/odbc.ini.

Next, edit /etc/cdr\_adaptive\_odbc.conf to include the following:

vi /etc/asterisk/cdr\_adaptive\_odbc.conf  
[asteriskcdr]  
connection=asterisk  
table=cdr  
alias start=calldate

Next, ensure CDR writing is enabled in cdr\_manager.conf:

vi /etc/asterisk/cdr\_manager.conf  
[general]  
enabled = yes

Finally, restart Asterisk:

sudo service asterisk restart

#### Node.js Integration

The CDR reporting functionality relies on two Node.js servers: the Management Portal (server-db.js) and the acr-cdr (app.js). These servers provide the following capabilities:

* server-db.js
  + Receives GET call for /cdrinfo.
  + Performs GET call to app.js /getallcdrrecs.
  + Acts as a middleman for the cdr.html page to the app.js node server. This node server can be bypassed by changing the cdr.html GET call from /cdrinfo to http://<app.js location>/getallcdrrecs if the CDR Dashboard needs to be separated from the Management Portal.
* app.js
  + Receives GET call for /getallcdrrecs.
  + Performs query to the Asterisk CDR database table. Returns a JavaScript Object Notation (JSON) object of the data.

## Agent/Agent Database (Provider)

ACE Direct uses a database to store Agent information and to verify Agent identity when an Agent logs into the system.

To host the Agent data, a MySQL database server is required. A RESTful API developed using Node.js provides access to the data for Agent verification. For this effort, the Health FFRDC built the Node.js server with the following software versions:

* Operating System: CentOS 7.x
* Node.js: v12.18.2

### MySQL Database Server Configuration

The Health FFRDC team developed a MySQL database containing several tables to store the Agent-related data. Create a database and use the dat/acedirectdefault.sql MySQL script found in <https://github.com/FCC> to create the application tables.

Verify that the MySQL database is up, running, and accepting connections. A tool like MySQL Workbench can quickly verify that the configuration is correct. The dat/acedirectdefault.sql script pre-populates the database with default agent data; however, you must first modify dat/acedirectdefault.sql to have actual values for the EXTENSION\_PASSWORD, \_ASTERISK\_PASSWORD\_, and \_ACEDIRECT\_PASSWORD\_ placeholders. Remember to update dat/config.json to include your actual database name, users, and passwords.

## Video Relay Service User Database (Provider)

The VRS user database was developed to emulate a VRS user lookup in the iTRS- User Registration Database (URD) from the Agent desktop until full access to the iTRS-URD is obtained. This lookup verifies that the Consumer-provided phone number is a registered VRS number. A MySQL database server is required. A RESTful API developed using Node.js provides access to the data for Consumer verification. For this effort, the Health FFRDC built the Node.js server with the following software versions:

* Operating System: CentOS 7.x or Amazon Linux 2
* Node.js: Version 12.18.2

### MySQL Database Server Configuration

The Health FFRDC team developed a MySQL database containing a single table to store the emulated VRS lookup data. The dat/acedirectdefault.sql script pre-populates the database with default VRS data.

Verify that the MySQL database is up, running, and accepting connections. A tool like MySQL Workbench can quickly verify that the configuration (username, password) is correct.

## STUN and TURN Server Installation

If a host is located behind a NAT firewall, it can be difficult (if not impossible) for that host to communicate directly with other hosts (peers). In these situations, the host must use the services of an intermediate node as a communication relay. This specification defines a protocol, called TURN (Traversal Using Relays around NAT), which allows the host to control the operation of the relay and to exchange packets with its peers using the relay. TURN differs from other relay control protocols because it allows a client to communicate with multiple peers using a single relay address. A TURN server, which is an implementation of the Session Traversal Utilities for NAT (STUN) protocol, uses a relay to provide an alternate method for NAT discovery and traversal (STUN). TURN can traverse symmetric NAT instances. The STUN server may be used by Asterisk if there is a Public Asterisk IP-Address; it may also be used by the Signaling-Server and the Kurento (KMS) server. In general, a TURN Server can run in STUN, TURN, or both modes. In practice, the STUN and TURN Servers are separated; this helps with debugging and prevents some security issues. For demonstration purposes, STUN is installed on Centos, and TURN is installed on Ubuntu.

### STUN Server Installation

For STUN services under CentOS, [turnserver](https://www.webrtc-experiment.com/docs/TURN-server-installation-guide.html#centos) will be used. Before installing turnserver, ensure the required dependencies are installed:

sudo yum -y install  
sudo yum install -y make gcc cc gcc-c++ wget openssl-devel libevent libevent-devel mysql-devel mysql-server

Then install the LibEvent modules, a dependency of turnserver:

wget https://github.com/downloads/libevent/libevent/libevent-2.0.21-stable.tar.gz

tar xvfz libevent-2.0.21-stable.tar.gz  
cd libevent-2.0.21-stable && ./configure  
sudo make && sudo make install && cd ..

Finally, install turnserver:

wget http://turnserver.open-sys.org/downloads/v3.2.3.8/turnserver-3.2.3.8.tar.gz  
tar -xvzf turnserver-3.2.3.8.tar.gz  
cd turnserver-3.2.3.8 && ./configure  
sudo make && sudo make install

If desired, you may configure a turnserver config file to add user credentials as well as the address and port to listen on:

mkdir /etc/turnserver  
vi /etc/turnserver/turnserver.conf  
   
# setting static accounts  
# Remember, "static" accounts are not dynamically checked by the turnserver process.  
user=username:password  
  
# listen ports  
listening-port=<port>  
listening-ip=<local\_ip>

Start the turnserver process. Use the first command to implement the config file, and the second not to:

nohup turnserver -v -r ip:port -z -c /etc/turnserver/turnserver.conf &  
nohup turnserver -L <local\_IP> -v -z --min-port 10000 --max-port 20000 -n &

#### STUN Server Startup

You can enable turnserver to start on system boot if desired. The following instructions have been validated on CentOS servers and may or may not work on other Linux distributions. To do so, first create the following script, make it executable, and save it as /etc/rc.d/init.d/turnserver:

#!/bin/bash  
# chkconfig: 2345 20 80  
# description: Manage turnserver as a system service so it starts on boot  
  
# Source function library  
. /etc/init.d/functions  
  
# This variable will be used to define what IP address turnserver listens on  
# You may need to change this depending on how your network service is configured  
HOST=$(hostname -I | awk '{print $1}')  
  
  
# This is the port that STUN will listen on.  
PORT=3478  
  
start() {  
 /usr/local/bin/turnserver -L $HOST -p $PORT -v -z --min-port 10000 --max-port 20000 -n  
 echo "STUN has started successfully"  
}  
  
stop() {  
 killall -9 turnserver  
 echo "STUN server stopped successfully"  
}  
  
case $1 in  
 start)  
 start  
 ;;  
 stop)  
 stop  
 ;;  
 restart)  
 stop  
 start  
 ;;  
 status)  
 STATUS=$(netstat -tanp | grep turnserver | grep $PORT)  
 if [ ${#STATUS} == 0 ]   
 then  
 echo "STUN server is not currently running"  
 else  
 echo "STUN server is currently running"  
 echo $STATUS  
 fi  
 ;;  
 \*)  
 echo "Usage: service turnserver (start|stop|restart|status)"  
 exit 1  
 ;;  
esac  
exit 0

Then, create the following systemd configuration file and save it as /usr/lib/systemd/system/turnserver.service:

[Unit]  
Description=Turnserver Service  
After=network-online.target  
Wants=network-online.target  
  
[Service]  
Type=simple  
ExecStart=/etc/rc.d/init.d/turnserver start  
;ExecStop=/root/start-turn.sh stop  
;ExecReload=/root/start-turn.sh restart  
  
[Install]  
WantedBy=default.target

Finally, reload the systemctl daemon, enable the turnserver service in systemctl, then reboot the server and use the 'netstat' command to confirm that turnserver is now starting upon boot:

systemctl daemon-reload  
systemctl enable turnserver.service  
reboot now

### TURN Server Installation

For TURN services on Ubuntu, coturn was used.

Run: apt-get update && apt-get install coturn

Edit: /etc/turnserver.conf and set the following variables (typical values are shown):

* listening-port (3478)
* listening-ip (private IP address)
* external-ip (public IP address)
* min-port (10000)
* max-port (20000)
* realm (domain name)
* user(user:password)

### TURN Server Startup

Edit /etc/init.d/coturn and add the following variables:

PATH=/usr/bin:/sbin:/usr/sbin:/bin

DESC=coturn # COTURN

NAME=coturn # TURN Server

PROCNAME=turnserver # Binary name

DAEMON=/usr/bin/turnserver

DAEMON\_ARGS="-c /etc/turnserver.conf -o -v" # Arguments to run the daemon with

PIDFILE\_DIR=/var/run

PIDFILE=/var/run/$PROCNAME.pid

SCRIPTNAME=/etc/init.d/$NAME

USER=turnserver

GROUP=turnserver

Edit /etc/default/coturn and set TURNSERVER\_ENABLED=1 (start on boot)

To start: turnserver –o –v, or run service start coturn

## iTRS ENUM Database

For phone numbers of d/Deaf or hard of hearing users, iTRS is the authoritative database. The lookup of the iTRS E.164 Number to URI Mapping (ENUM) database performs Domain Name System (DNS) queries to determine a Uniform Resource Identifier (URI) for a 10-digit telephone number. There is a GUI as well as a programmatic interface. **Note: Access to the iTRS ENUM database requires permission from the FCC.**

To query the production iTRS database requires establishing an Internet Protocol Secure (IPSec) tunnel with Neustar. For the proof of concept, the Health FFRDC team added the IP address <IP ADDRESS> first in the DNS settings (/etc/resolv.conf) for the ENUM lookup to work. Note that with the default Amazon Web Services (AWS) instance settings, any changes made to the resolv.conf file will be overwritten on restart of the network service because new values are queries from DNS. This behavior must be disabled.

The following example snippet of code, which is part of an Asterisk Dial Plan, comes from an extensions.conf file. Subsection 3.2.2.1 provides more information on the Asterisk Dial Plan.

exten => \_9.[1-9]XXXXXXXXXX, 1, Set(sipuri=${ENUMLOOKUP(+${EXTEN:1},sip,,1,itrs.us)})

same => n,NoOp("Outbound Direct Video Call to: ${EXTEN:1}") ; just for informational purposes

same => s,n,SipAddHeader(P-Asserted-Identity: <sip:nnnnnnnnnn>) ;set the callerID number

same => n,NoOp("sipuri: ${sipuri:1}")

same => n,Dial(SIP/${sipuri:1},30)

## strongSwan for Secure Socket Layer Tunnel

strongSwan is an open-source Virtual Private Network (VPN) software that is widely popular within the IPSec industry. strongSwan can be installed on both CentOS and Amazon Linux servers; the following instructions were implemented on an Amazon Linux EC2 instance.

In the following installation scenario, the local side of the VPN is running within AWS. The remote side provider requires both a public IP for the VPN endpoint and for the tunneled traffic. The implementation requires two public IP addresses on the local side and translates all local traffic to one of those addresses for the VPN tunnel. Also, on the remote end, there is only one accessible IP address. Minor adjustments would be required to allow access to more than one destination IP address or to allow a range of IP addresses directly (with or without NAT) through the VPN.

All the following commands must be run as root (or via sudo).

### strongSwan Installation

First, use yum to install strongSwan:

yum install -y strongswan

Then, modify the /etc/strongswan/ipsec.conf table to reflect the following:

config setup

#charondebug=”ike 2, net 3, knl 2, cfg 2” #useful debugs

conn %default

ikelifetime=480m

keylife=60m

rekeymargin=3m

keyingtries=1

keyexchange=ikev1

authby=secret

conn PeerProvider

auto=start

ike=aes128-sha1-modp1024 #P1: modp1536 = DH group 2

esp=aes128-sha1-modp1024 #P2

left=<strongSwan local address> #Local outside address

leftsubnet=<dummy address>/32 #network behind Local

leftid=<strongSwan public IP> #IKEID sent by Local

leftfirewall=yes

right=<remote peer address. #PeerProvider outside address

rightsubnet=<remote server address>/32 #network behind PeerProvider

ighted=<remote peer address. #IKEID sent by PeerProvider

Agree with your peer on a shared key and then modify the /etc/strongswan/ipsec.secrets file to reflect the following:

<strongSwan public IP> <remote server address> : PSK “<Key obtained from PeerProvider in quotes>”

IPTables must be configured to perform NAT between the source subnet and the public IP identified for tunnel traffic. Create the /etc/iptables.conf file and add the following:

\*nat

:PREROUTING ACCEPT [5:436]

:INPUT ACCEPT [1:92]

:OUTPUT ACCEPT [34:9996]

:POSTROUTING ACCEPT [34:9996]

-A POSTROUTING -s <VPC CIDR block>/24 -d <remote server address>/32 -o eth0 -j SNAT –to-source <dummy address>

COMMIT

\*filter

:INPUT ACCEPT [1063:95316]

:FORWARD ACCEPT [12:1032]

:OUTPUT ACCEPT [1018:375057]

COMMIT

The server needs a dummy interface associated with the public IP for tunnel traffic, and the IPTables configuration must be loaded at startup. Append the following lines to /etc/rc.d/rc.local:

/sbin/modprobe dummy

/sbin/ifconfig dummy0 <dummy address> netmask 255.255.255.0

/sbin/iptables-restore < /etc/iptables.conf

Move into the /etc/rc3.d directory and create the following symbolic links to ensure that strongSwan gracefully starts/stops when the EC2 instance is rebooted:

cd /etc/rc3.d

ln -s ../init.d/strongswan S48strongswan

ln -s ../init.d/strongswan K52strongswan

Routing must be enabled on the server. Modify the variable within the /etc/sysctl.conf file to the following:

net.ipv4.ip\_forward = 1

Create the following script in /root/scripts to automatically restart strongSwan if ITRS queries fail:

#!/bin/bash

DEBUG=true

LOG=/var/log/itrsmon.log

if ! dig @<remote server address> in naptr 0.9.8.7.6.5.4.3.2.1.1.itrs.us>/dev/null 2>&1

then

echo $(/bin/date) “ – ITRS Lookup failed” >> $LOG

/etc/init.d/strongswan restart

else

$DEBUG && echo $(/bin/date) “ – ITRS Lookup success” >> $LOG

fi

Add the following line to the crontab to monitor strongSwan by running the script once per minute:

$ crontab -e

\* \* \* \* \* /root/scripts/itrs\_mon.sh

Finally, reboot the instance to verify that the strongSwan service is running (using the ‘strongswan status’ command). Once the service is running, you can move onto the following “AWS Specific Config” section if you are in AWS. When you have completed those steps, your DNS queries to the iTRS database from Asterisk servers should be successful. An example of a successful iTRS query is as follows:

$ dig @<remote server address> in naptr 1.1.1.1.1.1.1.1.1.1.1.itrs.us

; <<>> DiG 9.9.4-RedHat-9.9.4-50.el7\_3.1 <<>> @<remote server address> in naptr 1.1.1.1.1.1.1.1.1.1.1.itrs.us

; (1 server found)

;; global options: +cmd

;; Got answer:

;; ->>HEADER<<- opcode: QUERY, status: NOERROR, id: 32364

;; flags: qr aa rd ad; QUERY: 1, ANSWER: 1, AUTHORITY: 0, ADDITIONAL: 1

;; WARNING: recursion requested but not available

;; OPT PSEUDOSECTION :

; EDNS : version : 0, flags :; udp : 5120

;; QUESTION SECTION :

;1.1.1.1.1.1.1.1.1.1.1.itrs.us. IN NAPTR

;; ANSWER SECTION:

1.1.1.1.1.1.1.1.1.1.1.itrs.us. 900 IN NAPTR 10 1 “u” “E2U+h323” “!^(.\*)$!h323:\\1@192.168.1.1!” .

;; Query time: 15 msec

;; SERVER: <remote server address>#53(<remote server address>)

;; WHEN: Fri Aug 04 14:40:59 UTC 2017

;; MSG SIZE rcvd: 115

### Amazon Web Services Environment Specific Configuration

#### Virtual Private Cloud Route Table

A route must be created in the virtual private cloud (VPC) route table to ensure that traffic to the iTRS database is routed to the VPN instance (which in turn routes it out over the VPN tunnel).

#### Source/Destination Checking

By default, instances drop packets when source and destination information do not match that of an instance. You can disable this behavior from the AWS console by selecting an instance and going to network options.

### strongSwan Troubleshooting

You can run ‘strongswan statusall’ to view the status of the VPN tunnel. The expected command output will look like the following:

$ strongswan statusall

Status of IKE charon daemon (strongSwan 5.4.0, Linux 4.9.38-16.33.amzn1.x86\_64, x86\_64):  
 uptime: 116 minutes, since Aug 04 12:46:06 2017  
 malloc: sbrk 1622016, mmap 0, used 502608, free 1119408  
 worker threads: 11 of 16 idle, 5/0/0/0 working, job queue: 0/0/0/0, scheduled: 2  
 loaded plugins: charon aes des rc2 sha2 sha1 md4 md5 random nonce x509 revocation constraints acert pubkey pkcs1 pkcs8 pkcs12 pgp dnskey sshkey pem openssl gcrypt fips-prf gmp xcbc cmac hmac ctr ccm gcm curl attr kernel-netlink resolve socket-default farp stroke vici updown eap-identity eap-md5 eap-gtc eap-mschapv2 eap-tls eap-ttls eap-peap xauth-generic xauth-eap xauth-pam xauth-noauth dhcp  
**Listening IP addresses:** <strongSwan local address>  
 <dummy address> **← both IP addresses should be listed**Connections:  
 PeerProvider: <strongSwan local address>…<remote peer address. IKEv1  
 PeerProvider: local: [<strongSwan public address>] uses pre-shared key authentication  
 PeerProvider: remote: [<remote peer address.] uses pre-shared key authentication  
 PeerProvider: child: <dummy address>/32 === <remote server address>/32 TUNNEL  
Security Associations (1 up, 0 connecting):  
 **PeerProvider[1]: ESTABLISHED 116 minutes ago**, <strongSwan local address>[<strongSwan public address>]...<remote peer address.[<remote peer address.]  
 PeerProvider[1]: IKEv1 SPIs: <<SPI>>, pre-shared key reauthentication in 21 hours  
 PeerProvider[1]: IKE proposal: AES\_CBC\_128/HMAC\_SHA1\_96/PRF\_HMAC\_SHA1/MODP\_1024  
 **PeerProvider{3}: INSTALLED, TUNNEL**, reqid 1, ESP SPIs: <<SPI>>  
 PeerProvider{3}: AES\_CBC\_128/HMAC\_SHA1\_96/MODP\_1024, 1494 bytes\_i (8 pkts, 2s ago), 611 bytes\_o (8 pkts, 2s ago), rekeying in 47 minutes  
 PeerProvider{3}: <dummy address>/32 === <remote server address>/32

To find system messages related to strongSwan, use the following grep command:

$ grep charon /var/log/messages  
**#Shutdown**  
Jul 27 00:27:58 ServerName charon: 00[DMN] signal of type SIGINT received. Shutting down  
Jul 27 00:27:58 ServerName charon: 00[IKE] closing CHILD\_SA PeerProvider{1} with SPIs <<SPI>> (0 bytes) 5401197d\_o (0 bytes) and TS <Dummy Address>/32 === <remote server address>/32  
Jul 27 00:27:58 ServerName charon: 00[IKE] sending DELETE for ESP CHILD\_SA with SPI cc95ab50  
Jul 27 00:27:58 ServerName charon: 00[ENC] generating INFORMATIONAL\_V1 request 4051018568 [ HASH D ]  
Jul 27 00:27:58 ServerName charon: 00[NET] sending packet: from <strongSwan public IP>[500] to <remote peer address.[500] (76 bytes)  
Jul 27 00:27:58 ServerName charon: 00[IKE] deleting IKE\_SA PeerProvider[1] between <strongSwan public IP>[<local server address>]...<remote peer address.[<remote peer address.]  
Jul 27 00:27:58 ServerName charon: 00[IKE] sending DELETE for IKE\_SA PeerProvider[1]  
Jul 27 00:27:58 ServerName charon: 00[ENC] generating INFORMATIONAL\_V1 request 3332107879 [ HASH D ]  
Jul 27 00:27:58 ServerName charon: 00[NET] sending packet: from <strongSwan public IP>[500] to <remote peer address.[500] (92 bytes)  
Jul 27 00:27:58 ServerName charon: 00[KNL] received netlink error: Address family not supported by protocol (97)

#Startup

Jul 27 00:27:58 ServerName charon: 00[DMN] Starting IKE charon daemon (strongSwan 5.4.0, Linux 4.9.32-15.41.amzn1.x86\_64, x86\_64)  
Jul 27 00:27:58 ServerName charon: 00[LIB] openssl FIPS mode(2) – enabled

#No IPv6 – that’s OK

Jul 27 00:27:58 ServerName charon: 00[NET] could not open socket: Address family not supported by protocol  
Jul 27 00:27:58 ServerName charon: 00[NET] could not open IPv6 socket, IPv6 disabled  
Jul 27 00:27:58 ServerName charon: 00[KNL] received netlink error: Address family not supported by protocol (97)  
Jul 27 00:27:58 ServerName charon: 00[KNL] unable to create IPv6 routing table rule

#No certs – that’s OK

Jul 27 00:27:58 ServerName charon: 00[CFG] loading ca certificates fr‘m '/etc/strongswan/ipsec.d/cace’ts'  
Jul 27 00:27:58 ServerName charon: 00[LIB] opening directo‘y '/etc/strongswan/ipsec.d/cace’ts' failed: No such file or directory  
Jul 27 00:27:58 ServerName charon: 00[CFG] reading directory failed  
Jul 27 00:27:58 ServerName charon: 00[CFG] loading aa certificates fr‘m '/etc/strongswan/ipsec.d/aace’ts'  
Jul 27 00:27:58 ServerName charon: 00[LIB] opening directo‘y '/etc/strongswan/ipsec.d/aace’ts' failed: No such file or directory  
Jul 27 00:27:58 ServerName charon: 00[CFG] reading directory failed  
Jul 27 00:27:58 ServerName charon: 00[CFG] loading ocsp signer certificates fr‘m '/etc/strongswan/ipsec.d/ocspce’ts'  
Jul 27 00:27:58 ServerName charon: 00[LIB] opening directo‘y '/etc/strongswan/ipsec.d/ocspce’ts' failed: No such file or directory  
Jul 27 00:27:58 ServerName charon: 00[CFG] reading directory failed  
Jul 27 00:27:58 ServerName charon: 00[CFG] loading attribute certificates fr‘m '/etc/strongswan/ipsec.d/ace’ts'  
Jul 27 00:27:58 ServerName charon: 00[LIB] opening directo‘y '/etc/strongswan/ipsec.d/ace’ts' failed: No such file or directory  
Jul 27 00:27:58 ServerName charon: 00[CFG] reading directory failed  
Jul 27 00:27:58 ServerName charon: 00[CFG] loading crls fr‘m '/etc/strongswan/ipsec.d/c’ls'  
Jul 27 00:27:58 ServerName charon: 00[LIB] opening directo‘y '/etc/strongswan/ipsec.d/c’ls' failed: No such file or directory  
Jul 27 00:27:58 ServerName charon: 00[CFG] reading directory failed

#Interesting stuff starts here – loading secrets

Jul 27 00:27:58 ServerName charon: 00[CFG] loading secrets fr‘m '/etc/strongswan/ipsec.secr’ts'  
Jul 27 00:27:58 ServerName charon: 00[CFG] **loaded IKE secret for <local server address> <remote server address>**  
Jul 27 00:27:58 ServerName charon: 00[LIB] loaded plugins: charon aes des rc2 sha2 sha1 md4 md5 random nonce x509 revocation constraints acert pubkey pkcs1 pkcs8 pkcs12 pgp dnskey sshkey pem openssl gcrypt fips-prf gmp xcbc cmac hmac ctr ccm gcm curl attr kernel-netlink resolve socket-default farp stroke vici updown eap-identity eap-md5 eap-gtc eap-mschapv2 eap-tls eap-ttls eap-peap xauth-generic xauth-eap xauth-pam xauth-noauth dhcp  
Jul 27 00:27:58 ServerName charon: 00[JOB] spawning 16 worker threads  
Jul 27 00:27:58 ServerName charon: 05[CFG] received stroke: add connecti‘n 'PeerProvi’er'  
Jul 27 00:27:58 ServerName charon: 05[CFG] added configurati‘n 'PeerProvi’er'  
Jul 27 00:27:58 ServerName charon: 09[CFG] received stroke: initia‘e 'PeerProvi’er'  
Jul 27 00:27:58 ServerName charon: 09[IKE] initiating Main Mode IKE\_SA PeerProvider[1] to <remote peer address.  
Jul 27 00:27:58 ServerName charon: 09[ENC] generating ID\_PROT request 0 [ SA V V V ]

#Sending and RECEIVING packets – good sign

Jul 27 00:27:58 ServerName charon: 09[NET] **sending packet**: from <strongSwan public IP>[500] to <remote peer address.[500] (228 bytes)  
Jul 27 00:27:58 ServerName charon: 15[NET] **received packet**: from <remote peer address.[500] to <strongSwan public IP>[500] (164 bytes)  
Jul 27 00:27:58 ServerName charon: 15[ENC] parsed ID\_PROT response 0 [ SA V V ]  
Jul 27 00:27:58 ServerName charon: 15[ENC] received unknown vendor ID: 05:16:dc:8a:88:2c:54:a5:66:90:dc:05:bd:da:3b:9e:c8:05:e5:86:12:00:00:00:1e:06:00:00  
Jul 27 00:27:58 ServerName charon: 15[IKE] received DPD vendor ID  
Jul 27 00:27:58 ServerName charon: 15[ENC] received unknown vendor ID: 48:65:61:72:74:42:65:61:74:5f:4e:6f:74:69:66:79:38:6b:01:00  
Jul 27 00:27:58 ServerName charon: 15[ENC] generating ID\_PROT request 0 [ KE No ]  
Jul 27 00:27:58 ServerName charon: 15[NET] sending packet: from <strongSwan public IP>[500] to <remote peer address.[500] (196 bytes)  
Jul 27 00:27:58 ServerName charon: 11[NET] received packet: from <remote peer address.[500] to <strongSwan public IP>[500] (196 bytes)  
Jul 27 00:27:58 ServerName charon: 11[ENC] parsed ID\_PROT response 0 [ KE No ]  
Jul 27 00:27:58 ServerName charon: 11[ENC] generating ID\_PROT request 0 [ ID HASH N(INITIAL\_CONTACT) ]  
Jul 27 00:27:58 ServerName charon: 11[NET] sending packet: from <strongSwan public IP>[500] to <remote peer address.[500] (108 bytes)  
Jul 27 00:27:58 ServerName charon: 07[NET] received packet: from <remote peer address.[500] to <strongSwan public IP>[500] (76 bytes)  
Jul 27 00:27:58 ServerName charon: 07[ENC] parsed ID\_PROT response 0 [ ID HASH ]  
**#Phase 1 (IKE) Established**Jul 27 00:27:58 ServerName charon: 07[IKE] **IKE\_SA** PeerProvider[1] **established b** <strongSwan public IP>[<local server address>]...<remote peer address.[<remote peer address.]  
Jul 27 00:27:58 ServerName charon: 07[IKE] scheduling reauthentication in 86056s  
Jul 27 00:27:58 ServerName charon: 07[IKE] maximum IKE\_SA lifetime 86236s  
Jul 27 00:27:58 ServerName charon: 07[ENC] generating QUICK\_MODE request 688951572 [ HASH SA No KE ID ]  
Jul 27 00:27:58 ServerName charon: 07[NET] sending packet: from <strongSwan public IP>[500] to <remote peer address.[500] (316 bytes)  
Jul 27 00:27:58 ServerName charon: 13[NET] received packet: from <remote peer address.[500] to <strongSwan public IP>[500] (316 bytes)  
Jul 27 00:27:58 ServerName charon: 13[ENC] parsed QUICK\_MODE response 688951572 [ HASH SA No KE ID ]  
**#Phase 2 (ISAKMP) Established**Jul 27 00:27:58 ServerName charon: 13[IKE] **CHILD\_SA** PeerProvider{1} **established** with SPIs X and TS <Dummy Address>/32 === <remote server address>/32  
Jul 27 00:27:58 ServerName charon: 13[ENC] generating QUICK\_MODE request 688951572 [ HASH ]  
Jul 27 00:27:58 ServerName charon: 13[NET] sending packet: from <strongSwan public IP>[500] to <remote peer address.[500] (60 bytes)

You can verify the proper configuration of the iptables rule to mask the source IP with the dummy address. The number of packets that have been applied to that rule should be displayed:

$ iptables -t nat -L -v

Chain PREROUTING (policy ACCEPT 804 packets, 66289 bytes)  
 pkts bytes target prot opt in out source destination  
  
Chain INPUT (policy ACCEPT 43 packets, 4165 bytes)  
 pkts bytes target prot opt in out source destination  
  
Chain OUTPUT (policy ACCEPT 51100 packets, 3555K bytes)  
 pkts bytes target prot opt in out source destination  
  
Chain POSTROUTING (policy ACCEPT 51076 packets, 3553K bytes)  
 pkts bytes target prot opt in out source destination  
 **785 63924 SNAT** all -- any eth0 <<local subnet>>/24 **<remote server address> to:<Dummy Address>**

You can use the 'ip route' command to ensure strongSwan has properly configured the route on the instance required to forward traffic into the VPN tunnel. The second and fourth lines of the following output are important:

$ ip route show table all  
default via …  
<<Remote Address>>/24 dev dummy0 proto kernel scope link src <Dummy Address>  
… dev eth0  
<local subnet> dev eth0 proto kernel scope link src <strongSwan public IP>  
broadcast .. dev dummy0 table local proto kernel scope link src <Dummy Address>  
<< deleted the rest of the output >>

## Commercial Customer Relationship Management

To demonstrate integration with a commercial Customer Relationship Management (CRM) service, ACE Direct connects to the Zendesk portal. ACE Direct sends JSON-based messages to the RESTful Zendesk API to manage and query customer records. Zendesk is a suite of web-based products that help companies provide better customer service. ACE Direct uses Zendesk to capture Consumer complaints. When a Consumer files a complaint, the ACE Direct software uses the Zendesk web service API to create and store a complaint ticket. This API also allows queries and updates of created tickets. When an ACE Direct Agent answers a Consumer call, the application requests the ticket information from Zendesk and displays it on the ACE Direct Agent Desktop portal.

A Zendesk account must be created to use the CRM demonstration feature.

## FenDesk Customer Relationship Management

FenDesk is a server that emulates the [Zendesk](https://www.zendesk.com/) ticketing system for ACE Direct or any client that needs a simple ticketing system. The software only implements the subset of Zendesk RESTful API calls that ACE Direct uses; however, it is expandable to include other API calls.

FenDesk uses a simple storage scheme. It creates, updates, and returns tickets as simple JSON text files. The filename for a ticket follows the same naming convention as Zendesk: <ticketno>.json (e.g., 322.json). FenDesk offers RESTful API calls to test connectivity, add/update/delete/retrieve tickets, and search for all tickets with a specified VRS number.

## Enterprise Service Bus

The enterprise service bus (ESB) is an optional component that provides a generic method to integrate with legacy database systems as well as the diverse number of databases and unstructured data repositories in use and on the market today.

### Background

Apache ServiceMix 6.1.2 is used as the service broker. Apache ServiceMix is an enterprise-class, open source, distributed ESB based on the service-oriented architecture (SOA) model. It is a project of the Apache Software Foundation and was built on the semantics and APIs of the Java Business Integration (JBI) specification JSR 208. The software is distributed under the Apache License.

The current version of ServiceMix fully supports the OSGi framework. ServiceMix is lightweight, easily embeddable, and has integrated Spring Framework support. It can be run at the edge of the network (inside a client or server), as a standalone ESB provider, or as a service within another ESB. ServiceMix is compatible with Java SE or a Java Enterprise Edition (EE) application server. ServiceMix uses ActiveMQ to provide remoting, clustering, reliability, and distributed failover. The basic frameworks used by ServiceMix are Spring and XBean.

ServiceMix comprises the latest versions of Apache ActiveMQ, Apache Camel, Apache CXF, and Apache Karaf. Additional installation features include:

* BPM engine via Activiti
* JPA support via Apache OpenJPA
* XA transaction management via JTA via Apache Aries

The ServiceMix ESB provides:

* Federation, clustering, and container-provided failover
* Hot deployment and life-cycle management of business objects
* Vendor independence from vendor-licensed products
* Compliance with the JBI specification JSR 208
* Compliance with the OSGi 4.2 specification through Apache Felix
* Support for OSGi Enterprise through Apache Aries

### Installation Overview

First install and configure ServiceMix and its prerequisites on the host machine.

#### ServiceMix System Requirements

To run Apache ServiceMix itself, you will need Java Runtime Environment (JRE) 1.8.x (Java 8), and about 100 MB of free disk space for the default assembly.

If you are developing your own integration applications and OSGi bundles, you will also need:

* Java Developer Kit (JDK) 1.8.x (Java 8)
* MySQL
* Apache Maven 3.0.4 or higher

The ACRDEMO broker application depends on MySQL for a database and Maven for building the application.

#### Installing the JDK

Issue the following commands to install the Java 8 JDK:

Red Hat/Fedora/CentOS Systems (SystemV):

sudo yum install java-1.8.0-openjdk

Set the JAVA\_HOME environment variable in the bash startup:

vi ~/.bashrc

Add the following lines to the end of the .bashrc script:

JAVA\_HOME= /opt/jdk1.8.0\_111

export JAVA\_HOME

JRE\_HOME=/opt/jdk1.8.0\_111/jre

export JRE\_HOME

PATH=$PATH:/opt/jdk1.8.0\_111/bin:/opt/jdk1.8.0\_111/jre/bin

export PATH

#### Installing Apache Maven

To install Apache Maven, issue the following command:

Redhat/Fedora/Centos Systems (SystemV):

sudo yum install maven

#### Installing MySQL

You will be able to set the password for the root account. **Note:** There is a current issue with the ESB that also requires setting the privileges for anonymous local users; accordingly, do not disable access for anonymous users. To install MySQL, issue the following commands:

Red Hat/Fedora/CentOS Systems (SystemV):

sudo yum install unixODBC unixODBC-devel libtool-ltdl libtool-ltdl-devel mysql-connector-odbc

wget http://repo.mysql.com/mysql-community-release-el7-5.noarch.rpm

sudo rpm -ivh mysql-community-release-el7-5.noarch.rpm

sudo yum install mysql-server

sudo systemctl start mysqld

sudo yum update

#### Configuring MySQL

The ServiceMix broker application for the Auto Call Routing (ACR) demo connects to the MySQL database; therefore, first create and configure the demo database.

Login to the MySQL command-line tool using the root account with the password you set earlier:

mysql -u root -p=somepassword

Create a database named “broker” for the ACR demo:

mysql> CREATE DATABASE broker;

mysql> USE broker;

Create the database user named “broker” for the ACR demo and set the password:

mysql> CREATE USER 'broker'@'localhost' IDENTIFIED BY 'somepassword';

Set the permissions for the database user “broker”. **Note:** This should be tuned to only grant the necessary privileges:

mysql> GRANT ALL PRIVILEGES ON broker.\* TO 'broker'@'%' WITH GRANT OPTION;

**Note:** There is a current issue with the ESB that also requires setting the privileges for anonymous local users. Privileges must be set for anonymous users:

mysql> GRANT ALL PRIVILEGES ON broker.\* TO ''@'localhost' WITH GRANT OPTION;

Create the “users” table:

mysql> CREATE TABLE `users` (

`user\_id` bigint(20) NOT NULL,

`user\_name` varchar(50) DEFAULT NULL,

`user\_description` varchar(45) DEFAULT NULL,

`user\_phone` varchar(20) DEFAULT NULL,

`user\_address` varchar(50) DEFAULT NULL,

`user\_account` varchar(50) DEFAULT NULL,

PRIMARY KEY (`user\_id`));

Populate the “users” table with test records. **Note:** The user\_id values need to correspond to IDs of Zendesk users:

mysql> INSERT INTO `users` VALUES

(3770168798,'John Doe ','Some Details','555-555-1111',NULL,'121212'),

(4060741111,'Jane Doe','No Description','222-111-1111','12341 Main Street','12345671'),

(4758821111,'Tim','No Description','555-666-7777','','5656565');

#### Downloading and Building Broker Application

Set up Secure Shell (SSH) keys to access the git repository according to these instructions:

* [Adding a new SSH key to your GitHub account](https://help.github.com/articles/adding-a-new-ssh-key-to-your-github-account/)

Create a folder for cloning the broker source code and navigate to that folder:

mkdir ~/code && cd ~/code

Clone the broker git repository from esb.git.

Navigate to the broker code folder:

cd camel-rest-proxy-blueprint/

Modify the applications blueprint file for your environment. The blueprint is configured to process messages intended for Zendesk. You will need to modify the blueprint to specify your Zendesk hostname and any proxy settings if you are accessing Zendesk from the ESB through a proxy.

Build the broker application with Maven:

mvn clean install

#### Downloading and Installing Apache ServiceMix

Apache ServiceMix 6.1.2 is available under the Apache License v2 and can be downloaded from <http://servicemix.apache.org/downloads/servicemix-6.1.2.html>.

Create and navigate to a folder where the downloaded zip file will be placed:

mkdir ~/dev-tools && cd ~/dev-tools

Download and uncompress the zip file. For example:

wget <http://mirror.cc.columbia.edu/pub/software/apache/servicemix/servicemix-6/6.1.2/apache-servicemix-6.1.2.zip>

unzip apache-servicemix-6.1.2.zip

#### Running and Configuring ServiceMix

In a command shell, navigate to the ServiceMix bin directory (e.g., ~/dev-tools/apache-servicemix-6.1.2):

cd ~/dev-tools/apache-servicemix-6.1.2/bin

Start ServiceMix:

./servicemix

Install the following features:

karaf@root>feature:install jdbc

karaf@root>feature:install pax-jdbc-mysql

karaf@root>feature:install camel-jsonpath

karaf@root>feature:install camel-jetty

karaf@root>feature:install camel-jdbc

karaf@root>feature:install camel-http4

You may need to download the pax-jdbc artifact from the Maven repository if the pax-jdbc-mysql install does not work:

karaf@root>feature:repo-add pax-jdbc 0.6.0

Create the Java Database Connectivity (JDBC) connection to the MySQL database:

**Note:** Depending on your version of jdbc, you will need either the command “jdbc:ds-create” or “jdbc:create”. Type <tab> to print a list of available commands and find the one you need:

karaf@root>jdbc:ds-create -dn mysql -url jdbc:mysql://localhost:3306/demo?user=broker&password=somepassword mySqlDataSource

karaf@root>jdbc:create -d mysql -t MySQL -url jdbc:mysql://localhost:3306/broker -u broker -p somepassword mySqlDataSource

Check that the JDBC connection was created:

karaf@root>jdbc:datasources

Another way to check the database connection is to issue a query:

karaf@root>jdbc:query jdbc/mySqlDataSource "select \* from users"

Install the broker application. The application will be installed as an OSGI bundle:

karaf@root>bundle:install -s mvn:org.apache.camel/camel-rest-proxy-blueprint/2.16.3

Check that the broker was installed. The bundle should be the last bundle in the list and its status should be ACTIVE:

karaf@root>bundle:list

#### Install and Start ServiceMix as a Service

Start the ServiceMix if is not already started. Issue the following commands:

karaf@root>feature:install wrapper

karaf@root>wrapper:install -s AUTO\_START -n KARAF -d Karaf -D “Karaf Service”

A message similar to the following will be displayed:

Setup complete. You may wish to tweak the JVM properties in the wrapper configuration file before installing and starting the service:

~/dev-tools/apache-servicemix-6.1.2/etc/KARAF-wrapper.conf

Redhat/Fedora/Centos Systems (SystemV):

To install the service:

$ ln -s ~//dev-tools/apache-servicemix-6.1.0/bin/KARAF-service /etc/init.d/

$ chkconfig KARAF-service --add

To start the service when the machine is rebooted:

$ chkconfig KARAF-service on

To disable starting the service when the machine is rebooted:

$ chkconfig KARAF-service off

To start the service:

$ service KARAF-service start

To stop the service:

$ service KARAF-service stop

To uninstall the service:

$ chkconfig KARAF-service --del

$ rm /etc/init.d/KARAF-service

For systemd compliant Linux:

To install the service (and enable at system boot):

$ systemctl enable~/dev-tools/apache-servicemix-6.1.2/bin/KARAF.service

To start the service:

$ systemctl start KARAF

To stop the service:

$ systemctl stop KARAF

To check the current service status:

$ systemctl status KARAF

To see service activity journal:

$ journalctl -u KARAF

To uninstall the service (and disable at system boot):

$ systemctl disable KARAF

Exit the ServiceMix/Karaf shell, by shutting down ServiceMix:

karaf@root>shutdown

Install the ServiceMix service:

sudo ln -s ~/dev-tools/apache-servicemix-6.1.2/bin/KARAF-service /etc/init.d/

Set the service to start when the machine is rebooted:

sudo update-rc.d KARAF-service defaults

Start the ServiceMix service:

sudo /etc/init.d/KARAF-service start

To log back into ServiceMix once the service is started, issue the following commands:

cd ~/dev-tools/apache-servicemix-6.1.2/bin

./client

To exit the ServiceMix shell without shutting down the service, type ^D (i.e., Ctrl-D). **Note:** If you type shutdown in ServiceMix shell, the entire service will be shut down.

### Editing blueprint.xml Application File

The blueprint.xml file is provided with placeholders that must be edited for your specific environment.

#### Update Zendesk Hostname

The blueprint.xml file has placeholders for the Zendesk hostname. You will need to provide your Zendesk hostname wherever you see the placeholder “<insert CRM hostname>”.

#### Enable/Disable Proxy

If there is a proxy between the ESB and Zendesk, you may need to add the following parameters to the set of parameters specified for each instance of the Zendesk endpoint. For example, you may need to replace:

/api?bridgeEndpoint=true&amp;throwExceptionOnFailure=false

with:

/api?bridgeEndpoint=true&amp;proxyAuthHost=<replace with proxy ip>&amp;proxyAuthPort=<replace with proxy port>&amp;proxyAuthScheme=http4&amp; throwExceptionOnFailure=false

substituting your proxy host and port settings for the placeholders.

After editing the blueprint file, rebuild the application using Maven:

cd ~/code/camel-rest-proxy-blue-print/

mvn clean install

cd ~/dev-tools/apache-servicemix-6.1.2/bin

./client

karaf@root>bundle:install -s mvn:org.apache.camel/camel-rest-proxy-blueprint/2.16.3

Make sure the bundle is successfully installed with no errors and active.

### Testing the Broker Application

At this point, all of the code for the Broker application is contained in the OSGI Blueprint file (i.e., blueprint.xml), which can be viewed in the GitHub repository <https://github.com/FCC>.

If the Broker application is running in ServiceMix running, you can check the application by using curl at the command line. For example:

$ curl -u username@hostname/token:hLLUnPzJtpvMZ5WnntN3wCneKHkl20kP0Hhn5NrD http://localhost:9090/api/v2/users/me.json --insecure

where username is a Zendesk user account and hostname is your Zendesk host name.

You can check the status and statistics of the main Broker route in the ServiceMix/Karaf shell:

karaf@root>camel:route-info rest-http-zendesk-mysql-demo

Here is example output from the camel:route-info command:

Camel Route rest-http-zendesk-mysql-demo  
Camel Context: camel-1  
State: Started  
State: Started  
  
Statistics  
Exchanges Total: 2  
Exchanges Completed: 2  
Exchanges Failed: 0  
Exchanges Inflight: 0  
Min Processing Time: 240 ms  
Max Processing Time: 494 ms  
Mean Processing Time: 367 ms  
Total Processing Time: 734 ms  
Last Processing Time: 240 ms  
 Delta Processing Time: -254 ms  
Start Statistics Date: 2016-07-19 14:43:44  
Reset Statistics Date: 2016-07-19 14:43:44  
First Exchange Date: 2016-07-19 15:12:15  
Last Exchange Date: 2016-07-19 15:12:3

## SIP Proxy Server – Kamailio

### Introduction

Kamailio is a free, open-source software framework licensed under GPL that functions as a SIP Proxy Server able to handle thousands of call setups per second. Kamailio has been adopted and integrated into ACE Direct to build a large-scale, full-duplex streaming video and voice call center for real-time communications. It is featured with capabilities to support WebRTC-based Voice Over Internet Protocol (VoIP) (video and voice) calls; presence; Real-Time Text (RTT), i.e., Instant Messaging; and videomail. Moreover, it can be easily used for scaling up SIP-to-Public Switch Telephone Network (PSTN) gateways, PBX systems (such as Asterisk), and media servers.

### Installation and Configuration

Kamailio can be installed on a CentOS 7+ or an Amazon Linux 2 server.

The main install script (AD\_kamailio-install.sh) will install and configure Kamailio for ACE Direct. This script, which is available at <https://github.com/FCC>, can be used to quickly install and configure the SIP Proxy Server and the supporting RTP Proxy Server (rtpengine). This installation process will be facilitated by running the main script in conjunction with the supporting scripts, SQL, and oth configuration files in this same folder.

Generally, the process will be straightforward to start and streamlined by running the main shell script at the directory where it is located, as shown, using “default” configuration values when prompted during the execution:

$ sudo ./AD\_kamailio-install.sh

The user will need to provide the IP address of the Asterisk Server. The script will prompt for the other required fields and provide default values.

Default values have been used for the ongoing ACE Direct lab-based integration and testing. For a new ACE Direct installation and configuration on different deployment environments, new values should be used by the installers for their specific deployment.

## Kurento Media Server

### Introduction

Kurento is a WebRTC media server offering a set of client APIs to provide advanced media processing capabilities. Kurento has been adopted into ACE Direct to provide transcoding, multi-party communications, call recording, call monitoring, and bandwidth controls.

### Kurento Media Server Installation and Configuration

The Kurento media server requires an Ubuntu Linux 18.04 LTS (or newer) host. Follow these steps to perform the initial installation.

## Update \*apt\* and install \*kurento\*

sudo apt-get update

## Import the key from Kurento TEAM

sudo apt-key adv --keyserver keyserver.ubuntu.com --recv-keys 5AFA7A83

## Set DISTRO to "bionic" for Ubuntu 18.04, "xenial" for Ubuntu 16.04

DISTRO="bionic"

## Adding repo to source

sudo tee "/etc/apt/sources.list.d/kurento.list" >/dev/null << EOF

# Kurento Media Server - Release packages

deb [arch=amd64] http://ubuntu.openvidu.io/6.11.0 $DISTRO kms6

EOF

unset DISTRO

## Install kurento-media-server

sudo apt-get update && sudo apt-get install --yes kurento-media-server

The Kurento media server uses a combination certificate that combines the private key, certificate, and intermediate CA:

cat key.pem fullchain.pem > server.pem

Once the Kurento media server is installed, configuration files need to be modified.

* /etc/kurento/kurento.conf.json
  + The main configuration file to provide settings of Kurento Media Server
  + Defines the WS/WSS ports and path to combo certificates
  + Please refer to this [link](https://doc-kurento.readthedocs.io/en/6.9.0/features/security.html#configure-kurento-media-server-to-use-secure-websocket-wss) for more details
* /etc/kurento/modules/kurento/WebRtcEndpoint.conf.ini
  + Specific parameters for WebRtcEndpoint
  + Please refer to the repository directory in acedirect-kurento/confs/kurento/ examples
* /etc/kurento/modules/kurento/MediaElement.conf.ini
  + Generic parameters for all types of MediaElements
  + Please refer to the [Kurento GitHub](https://github.com/Kurento/kms-core/tree/6.11.0/src/server/config) page for examples
* /etc/kurento/modules/kurento/SdpEndpoint.conf.ini
  + Audio/video parameters for SdpEndpoints (i.e., WebRtcEndpoint and RtpEndpoint)
  + Please refer to the Kurento GitHub page for examples
* /etc/kurento/modules/kurento/HttpEndpoint.conf.ini
  + Specific parameters for HttpEndpoint
  + Please refer to the [Kurento GitHub](https://github.com/Kurento/kms-core/tree/6.11.0/src/server/config) page for examples

## Busylight Installation

### Installation

Complete installation instructions for the BusyLight server software are in the software repository in the ace-direct/obusylight folder.

### Using the Busylight with ACE Direct

1. Plug the *Kuando Busylight* into an available USB port on your computer. The light will flash red, green, blue twice, then turn off.
2. Double-click the lightserver.jar file. If prompted, open the file as a *Java(TM) Platform SE binary*.
3. The *Busylight - ACE Direct* UI will start, and the Busylight device will flash rainbow colors and turn off.
4. From a browser, log into ACE Direct.

* If prompted, allow the browser to use your camera.
* A Busylight dialog will appear. Click the Test Busylight Connection button.
* A new browser tab will appear. Click Advanced and click Proceed to localhost (unsafe).
* Busylight access will be granted. Close this tab.
* From the Agent Portal, close the Busylight dialog.

You are now ready to use the Busylight device with ACE Direct.

## ACE Quill Service (Call Captioning and Translation)

### Introduction

The ACE Quill Service is a Node.js server that provides call captioning to ACE Direct and a RESTful API for language translation.

### Installation and Configuration

The ACE Quill Service should be installed on the Asterisk server. The prerequisites for the ACE Quill Service are:

* Node.js
  + Version 12.18.2

To install the ACE Quill Service, create a Google Cloud Platform or IBM Cloud Account.

Google Cloud Platform:

1. The ACE Quill service uses the Google speech to text engine to support captions and requires a Google account
2. Create a Google account and create a speech to text service
3. Download a private key as a JSON from the Service Accounts menu
4. Copy the JSON file to config/google.json

IBM Cloud:

1. Configure ACE Quill service to use IBM Watson in place of Google for Speech to Text and Translation. Change the following files:
   1. service.js: uncomment line 3 and comment out line 4.
   2. api/models/acequillModel.js: uncomment line 1 and comment out line 2.
2. Create an IBM Watson Service Account
   1. The ACE Quill service uses the IBM Watson speech to text engine to support captions and requires an [IBM Cloud](https://www.ibm.com/cloud) account.
   2. Create an IBM Watson account and create a speech to text resource.
   3. Download the credentials file which should have the following format:  
        
      SPEECH\_TO\_TEXT\_IAM\_APIKEY=<API KEY HERE>  
      SPEECH\_TO\_TEXT\_URL=<WATSON URL HERE>
   4. Copy the API key and Universal Resource Locator (URL) into config/watson.js

In the ace-direct/dat/config.json, there is a translation\_server section that needs to be configured with the private IP address of the ACE Quill server: An example of that section of the configuration file is shown below:

"translation\_server": {  
 "private\_ip": "private IP address of ACE Quill server",  
 "port": "8005",  
 "protocol": "http",  
 "enabled": "true | false"  
 },

To start the ACE Quill node server:

1. Clone the ACE Quill service repository to the Asterisk server
2. Change directory to the `acequill-service`
3. Run `npm install pm2 -g`
4. Run `npm install`
5. Run `pm2 start process.json`

## ClamAV Service Configuration

ClamAV is an open-source antivirus engine that ACE Direct uses to scan files that are shared by consumers or agents. It runs as a service on the server and ACE Direct uses a Node.js library to execute it on uploaded files.

To install ClamAV itself on a CentOS system, try:

# yum install epel-release

# yum -y install clamav-server clamav-data clamav-update clamav-filesystem clamav clamav-scanner-systemd clamav-devel clamav-lib clamav-server-systemd

ClamAV uses virus definition files, which need to be kept up to date. The freshclam utility is used to update the virus definition files and it should be run once a day. If it is run too often, your server may be rate-limited.

To run freshclam manually, try:

# sudo -E freshclam

It is best to set up a cron job or similar to run freshclam every day. The install process should have created a cronjob in /etc/cron.d/clamav-update but you may have to update it to include an http\_proxy variable if necessary.

An example clamav-update cron file:

[MAILTO=admin@yourdomain.org](mailto:MAILTO=admin@yourdomain.org)

https\_proxy=http://<Proxy IP address>:<Proxy port> 0 \* \* \* root /usr/bin/freshclam >> /home/someuser/freshclam-output.log 2>&1

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# Conclusion

ACE Direct is a large system of simple and complicated components. This installation guide simplifies the installation, configuration, and deployment of ACE Direct. Where appropriate, this guide refers to other helpful documents online and in the GitHub repositories.

For more information or to post an issue or question, visit the FCC ACE Direct GitHub repo (<https://github.com/FCC>).

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Acronyms

| Term | Definition |
| --- | --- |
| ACE | Accessible Communications for Everyone |
| ACR | Auto Call Routing |
| ADA | Americans with Disabilities Act |
| AMI | Asterisk Management Interface |
| API | Application Programming Interface |
| ASL | American Sign Language |
| AWS | Amazon Web Services |
| BI | Business Intelligence |
| CA | Communication Assistant, Certificate Authority |
| CDR | Call Detail Record |
| COE | Center of Expertise |
| CMS | Centers for Medicare & Medicaid Services |
| CRM | Customer Relationship Management |
| DNS | Domain Name System |
| DVC | Direct Video Calling |
| EE | Enterprise Edition |
| ENUM | E.164 Number to URI Mapping |
| ESB | Enterprise Service Bus |
| FCC | Federal Communications Commission |
| FQDN | Fully Qualified Domain Name |
| FFRDC | Federally Funded Research and Development Center |
| GUI | Graphical User Interface |
| HHS | Department of Health and Human Services |
| HSTS | HTTP Strict Transport Security |
| HTTPS | HyperText Transport Protocol Secure |
| iTRS | Internet Telecommunications Relay Service |
| IP | Internet Protocol |
| IPSec | Internet Protocol Secure |
| JBI | Java Business Integration |
| JDBC | Java Database Connectivity |
| JRE | Java Runtime Environment |
| JSON | JavaScript Object Notation |
| NAT | Network Address Translation |
| ODBC | Open Database Connectivity |
| **OpenAM** | Open Access Management |
| OS | Operating System |
| OSGi | OSGi Alliance (formerly Open Systems Group Initiative) |
| PBX | Private Branch Exchange |
| POC | Proof of Concept |
| PSTN | Public Switch Telephone Network |
| REST | Representational State Transfer |
| RTT | Real-Time Text |
| **SIP** | Session Initiation Protocol |
| SOA | Service-Oriented Architecture |
| SQL | Structured Query Language |
| SSH | Secure Shell |
| SSL | Secure Socket Layer |
| STUN | Session Traversal Utilities for NAT |
| TCP | Transmission Control Protocol |
| TLS | Transport Layer Security |
| TRS | Telecommunications Relay Service |
| TURN | Traversal Using Relay NAT |
| UDP | User Datagram Protocol |
| UI | User Interface |
| URD | User Registration Database |
| URI | Uniform Resource Identifier |
| URL | Universal Resource Locator |
| VoIP | Voice Over Internet Protocol |
| VPC | Amazon’s Virtual Private Cloud |
| VPN | Virtual Private Network |
| VRS | Video Relay Service |
| WebRTC | Web Real-Time Communication |
| WS | WebSocket |
| WSS | WebSocket Secure |

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1. <https://www.fcc.gov/ace> [↑](#footnote-ref-2)