Quality Management Suite

Supported PBX Guide

Document Reference
QMS-PBX-Guide-Dec2017

Enghouse Interactive
Trevor Davis

1 Change control

Version	Author	Date	Comments
0.1	Trevor Davis	24 Nov 2016	Draft for internal review
1.0	Trevor Davis	30 Jan 2017	Approved release
1.1	Trevor Davis	21 Feb 2017	Cisco UCM 11.5 testing notice
1.2	Trevor Davis	11 Dec 2017	V7.1 Release

Contents

2 Introduction

Enghouse Interactive delivers technology and expertise to maximize the value of every customer interaction. The Company develops the world's most comprehensive portfolio of interaction management solutions, spanning structured, unstructured and self-service interactions. Core technologies include contact center, attendant console, IVR, call recording and quality management solutions that support many telephony environments, either on premise or in the Cloud. Enghouse Interactive has thousands of customers worldwide, supported by a global network of partners and more than 1000 dedicated staff across the company's 16 international operations.

2.1 Document Purpose

The following document provides details of each of the PBX supported with the QMS 7.1 release.

2.2 Quality Management Suite (QMS) Overview

QMS Call Recording is a VoIP call recording and monitoring system for IP or IP-enabled telephone systems. The QMS recording software records and monitors telephone calls from anywhere on the network and runs on Microsoft Windows servers (physical or virtual). The software enables businesses to obtain greater compliance and insight into their business operations through on-demand and always-on recording, silent monitoring, screen capture tools, a quality management facility for agents/operators, real-time speech analysis and agent coaching, and provides integration with back office business applications. The architecture of the QMS software ensures low cost of ownership, both during run time and future upgrades.

The Quality Management Suite includes the following components:

- **Call Recording core component** - Use for audio recording and monitoring software;
- **Agent Evaluation optional component** - Use for call scoring and agent coaching;
- **Screen Recording optional component** - Use for desktop screen recording and monitoring software;
- **Text Recording optional component** – Use for recording text-based communications such as email, instant messages, and webchat.
- **Real-time Speech Analytics optional component** – QMS is integrated with VocalCoach, which provides a real-time agent coaching application that utilises speech recognition and speech analysis algorithms to provide real-time feedback to agents whilst calls are in progress.

2.3 Supported PBX Platforms

The QMS suite is tested alongside a number of PBX platforms. As well as the following specific PBX list a number of additional PBX platforms can be supported using our generic SIP interface. Unfortunately, with our SIP integration, we are not in a position to test all of these environments and therefore cannot provide any guarantees on the level of functionality provided.

The PBX platforms formally supported are:

- Avaya – Aura Communication Manager, Communication Server 1000 and IP Office;
- Cisco – Unified Communications Manager, Unified Communications Manager Express and Business Edition 6000;
- Innovaphone
- Microsoft Skype for Business (formerly Lync);
- Mitel 3300;
- NEC – 8100, 8300, 8500, 9100, 9300, 9500, 3C (SIP);
- ShoreTel.

3 QMS PBX Requirements

QMS employs a distributed-services software architecture that is ideal for single and multiple-site deployments. The QMS solution is scalable to thousands of users and can be managed from any location via the Web-based QMS client interface. Flexible storage options allow administrators to store calls on local servers or on network storage locations.

The QMS server will always need to be provided with two sets of data; the audio traffic to be recorded and signalling/call control information. This data will always be routed to the Recording Service servers. A variety of common tools are available to deliver this data to the recording service as well as some PBX specific tools. The method used to deliver this data will have a direct impact on the features that can be supported on the QMS system and on the architecture of the solution.

Please be aware that QMS has no direct control over these two data sources and it is therefore the responsibility of the telephony/network architect to ensure that these two information streams are available to the QMS server.

3.1 Call Control Information

Signalling/call control information is provided via a Computer Telephony Interface (CTI), which will be provided by and will be specific to the PBX deployed. CTI components are PBX specific:

- For Avaya Communications Manager, TSAPI or DMCC is required;
- For Avaya CS1000, Meridian Link Services (MLS) is required;
- For Avaya IP Office, TAPI is required;
- For Cisco, TAPI is required when forked audio is being used (UCM systems only), with Skinny (SCCP) no CTI interface is required as the call data is delivered with the SCCP packets;
- For Innovaphone, the Innovaphone TAPI interface is required for call control data;
- For Mitel, integrations, the Mitel MiTAI interface is required;
- For Microsoft Skype for Business (formerly Lync), additional components and configurations are required;
- For NEC, CTI events are captured using port mirror on the network switch;
- For ShoreTel, ShoreTel TAPI is required.

For more information on PBX integrations, refer to the Quality Management Suite 7.1 Integration Guides.

3.2 Audio Streams

The ability to deliver audio streams to the QMS Recording Service is dependent on the tools used in the PBX/network deployment. For example, some network switches are capable of sending duplicate/multiple spans and therefore a span can be sent to multiple recording systems, some tools can only provide a single audio stream and therefore duplicate recording solutions are very difficult to deliver. Please engage a qualified design engineer to ensure that the deployed architecture will provide the recording feature required.

4 Supported PBX Matrix

SUMMARY MATRIX	Avaya Aura DMCC	Avaya IP Office	Avaya CS1000	Cisco CUCM	Cisco Forked Audio	Innovaphone	Microsoft Skype for Business	Mitel	Mitel SRC	NEC SV8100	NEC SV8300/SV8500	ShoreTel	Generic SIP
Internal Call Recording	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Call Chaining	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
User Mobility	✓	✓	✓	✓	✓		✓		✓	✓	✓	✓	✓
DNIS Support				✓				✓	✓			✓	
Concurrent Call Recording	✓	✓	✓	✓	✓	✓	✓	✓	✓			✓	✓
PBX Side Call Encryption					✓		✓		✓				
Phone Initiated Recording				✓	✓								
Dual Endpoint Recording				✓									
Private Line Appearances				✓									
Port Mirroring Required		✓	✓	✓		✓		✓		✓	✓	✓	✓
CTI System Required	✓	✓	✓		✓	✓		✓	✓			✓	
CT Connect Required			✓										
Additional Licensing Required	✓	✓	✓	✓				✓	✓			✓	
Vendor Certified				✓	✓		✓					✓	
QMS High Availability Supported				✓	✓		✓			✓	✓		✓
Speech Analytics Supported		✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓

5 PBX Summary Details

5.1 Avaya Aura with DMCC

Avaya Aura with DMCC	
Last Version Tested:	QMS 7.1 was tested against Avaya AES 6.1
Special Features:	None
Special Configuration Needed:	None
Limitations:	None
Licensing:	Customers will require an Avaya AES server along with one TSAPI Basic license and one DMCC license per user
Technical Details:	Audio Source: Avaya DMCC API Call Control Source: TSAPI / Avaya AES

5.2 Avaya IP Office

Avaya IP Office	
Last Version Tested:	QMS 7.1 was tested against Avaya IP Office R10
Special Features:	None
Special Configuration Needed:	SNMP must be enabled on the PBX and PBX user ID's must be specified in the QMS user configuration in order for user mobility to work. See Integration Guide for more details.
Limitations:	None
Licensing:	One Avata CTI Link Pro license required per PBX
Technical Details:	Audio Source: Port mirroring Call Control Source: TAPI

5.3 Avaya (Nortel) CS1000

Avaya (Nortel) CS1000	
Last Version Tested:	QMS 7.1 was tested against Avaya CS1000 version 4.5.
Special Features:	QMS can operate in contact center mode with Symposium to allow for agent mobility. Special configuration is required.
Special Configuration Needed:	MLS (Meridian Link Services) is required. CTI Connect 8.1 is also required and must be configured to operate with MLS. License CTI Connect license - CTC80-MOUNL is required. This provides support for unlimited users.
Limitations:	None
Licensing:	MLS must be purchased and installed in advance. CTI Connect 8.1 must be added to the order and licensed. Requires AACC.
Technical Details:	Audio Source: Port mirroring Call Control Source: CTI Connect and MLS

5.4 Cisco Unified Communications Manager

Cisco Unified Communications Manager	
Last Version Tested:	QMS 7.1 was tested against CUCM 11.5.
Special Features:	There are two features unique to this integration: Dual endpoint recording and private line appearances. Dual endpoint recording allows the user to share a primary extension between both a hard phone and a soft phone. Private line appearances allows the user to designate one line appearance as "private" so that it will not be recorded. This does not impact recording for other lines on the phone. A Cisco phone app is available for purchase and customization that will allow the user to pause/resume/start/stop recording from their phone. A virtual machine OVA template is available for deployment on Cisco BE6000 servers
Special Configuration Needed:	None
Limitations:	Multi-port ATA proxies are not supported. Single-port ATA devices are supported. If multi-port ATA's are required, consider using forked audio or a hybrid installation. Audio encryption is listed as supported but there are no current deployments using it. It is very complicated to set up and has special requirements.
Licensing:	Customer must pay SCCP licensing fee when SCCP is deployed. SIP phones do not require SCCP license.
Technical Details:	Audio Source: Port mirroring Call Control Source: Port mirroring of either SIP or SCCP packets Cisco UC5xx series and Cisco Unified Communications Manager Express are supported If both SIP and SCCP endpoints are being recorded, two recording servers are required.

5.5 Cisco Unified Communications Manager with Forked Audio (dual media streaming)

Cisco Unified Communications Manager with Forked Audio (dual media streaming)	
Last Version Tested:	QMS 7.1 was tested against CUCM 11.5.
Special Features:	A Cisco phone app is available for purchase and customization that will allow the user to pause/resume/start/stop recording from their phone. **/## is supported for starting and stopping recording from the handset if the handsets are SCCP/Skinny, SIP handsets do not support this feature.
Special Configuration Needed:	Must be configured in the PBX. See integration guide for more details.
Limitations:	Phones and other devices must support Built-in-bridge/dual media streaming CUCM version 7 or greater required Cisco UC5xx series and Cisco Unified Communications Manager Express are NOT supported
Licensing:	None
Technical Details:	Audio Source: Built-in-bridge on phone Call Control Source: TAPI

5.6 Innovaphone

Innovaphone	
Last Version Tested:	QMS 7.1 was tested against Innovaphone 11R2.
Special Features:	
Special Configuration Needed:	Requires Innovaphone TAPI.
Limitations:	
Licensing:	
Technical Details:	Audio source: Port mirroring Call control: TAPI

5.7 Microsoft Skype for Business with Network Packet Capture and/or RTPDataCollector on Clients

Microsoft Skype for Business with Network Packet Capture and/or RTPDataCollector on Clients	
Last Version Tested:	QMS 7.1 was tested against Skype for Business Server 2015
Special Features:	Microsoft RTAudio and Siren7 codecs are supported. Different types of calls can be recorded depending on which type of integration is desired.
Special Configuration Needed:	Requires software installation on Skype for Business Front-End Server.
Limitations:	Caller ID and dialed digits will appear in SIP URI format
Licensing:	None
Technical Details:	Audio Source: Port mirroring or RTPDataCollector. RTPDataCollector can run on client PC's, mediation servers, or edge servers depending on desired configuration. Refer to the Skype for Business integration guide. Call Control Source: Front End Server Plugin Many plug-in USB hardware devices such as headsets and speaker phones are supported.

5.8 Mitel 3300

Mitel 3300	
Last Version Tested:	QMS 7.1 was tested against MiVoice 8.0 PR2
Special Features:	None
Special Configuration Needed:	Encryption must be disabled.
Limitations:	Mitel 200ICP is no longer supported. Mitel ACD is supported only if user does not change phone devices. Teleworker users are not supported.
Licensing:	MiTAI license is required.
Technical Details:	Audio Source: Port mirroring Call Control Source: MiTAI

5.9 Mitel 3300 with Secure Recording Connector

Mitel 3300 with Secure Recording Connector	
Last Version Tested:	QMS 7.1 was tested against Mitel SRC v7.1.45.0
Special Features:	Redundant SRC servers are now supported. Audio between QMS and SRC are now encrypted.
Special Configuration Needed:	Mitel SRC server is required. Users must be configured to have calls go through the SRC server
Limitations:	None
Licensing:	MiTAI license is required. SRC server is required.
Technical Details:	Audio Source: Mitel SRC Call Control Source: MiTAI

5.10NEC SV8100/9100

NEC SV8100/9100	
Last Version Tested:	QMS 7.1 was tested against NEC SV9100
Special Features:	None
Special Configuration Needed:	None
Limitations:	None
Licensing:	None
Technical Details:	Audio Source: Port mirroring Call Control Source: Port mirroring of N-SIP packets

5.11NEC SV8300/8500/9300/9500

NEC SV8300/SV8500/SV9300/SV9500	
Last Version Tested:	QMS 7.1 was tested against NEC SV9300
Special Features:	Both Dterms (PROTIMS) and N-SIP handsets are supported
Special Configuration Needed:	None
Limitations:	None
Licensing:	None
Technical Details:	Audio Source: Port mirroring Call Control Source: Port mirroring of N-SIP and Protims packets

5.12ShoreTel

ShoreTel	
Last Version Tested:	QMS 7.1 was tested on ShoreTel 14.2
Special Features:	None
Special Configuration Needed:	Requires a Windows Server Platform.
Limitations:	Windows Server 2012 R2 is not supported by the ShoreTel TSP. Windows Server 2012 is supported, however.
Licensing:	For v6, a ShoreTel DVS license is required for $995 (ShoreTel part #21020). For v7 and higher, a ShoreTel TAPI application license is required for $495 (ShoreTel part #30049) or a DVS license
Technical Details:	Audio Source: Port mirroring Call Control Source: TAPI

5.13Generic SIP

Generic SIP	
Last Version Tested:	N/A
Special Features:	Known supported SIP PBX's in production: 3Com VCX, Asterisk, Broadsoft, Fonality, NEC 3C, Whaleback (not pure SIP), Zultys Many more are believed to be in production.
Special Configuration Needed:	None
Limitations:	The PBX and phones must implement standard SIP protocol.
Licensing:	None
Technical Details:	Audio Source: Port mirroring Call Control Source: Port mirroring of SIP packets