



Avaya Solution & Interoperability Test Lab

Application Notes for FROX AG talkbase 22.02 with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1– Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate FROX AG talkbase with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. FROX AG talkbase is an IP Attendant that integrates with Avaya Aura® Session Manager using both a SIP trunk connection and a SIP user connection.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required to integrate FROX AG talkbase with Avaya Aura® Communication Manager R10.1 and Avaya Aura® Session Manager R10.1. FROX AG talkbase is an IP Attendant that integrates with Avaya Aura® Session Manager using both a SIP trunk connection and a SIP user connection.

FROX AG talkbase has a web-based attendant console, talkbase Attendant, which works with a number of telephony platforms including Communication Manager and Session Manager. It interfaces with Active Directory, Presence and Exchange servers. Compliance testing focused on telephony functionality only and Presence or Exchange server functionality was not covered. Interoperability with Avaya H.323 endpoints in the Avaya environment is not supported or tested.

2. General Test Approach and Test Results

The general test approach evaluated the ability of talkbase Attendant to place calls and to receive calls placed to the main number routed to an available attendant.

talkbase must reside on a domain with an Active Directory controller and it requires a separate server within that domain. talkbase relies on the user management of Active Directory. One user account is required per attendant, with one Avaya SIP user (assigned an extension and password). talkbase internally queues main number calls to attendants using WebRTC connections and attendant consoles answer these calls. talkbase will not call attendants directly so they will not receive calls made to their number.

Because talkbase creates a self-signed certificate and installs it, both the clients and the talkbase server must be on the same domain as the Active Directory controller otherwise there may be an issue with the trust of the certificates issued.

A SIP user was created for each talkbase Attendant user. talkbase server registers the user to Session Manager when talkbase Attendant logs in. Please note that Application Sequences must not be defined on these users. A dialplan was added to route calls to talkbase Attendant. This would typically be the “main number” of the company and calls to that number would be answered by one of the attendants.

Calls were placed to the “main number” which is configured on Communication Manager to route across the SIP trunk to the talkbase server. Any calls then made to the main number were routed to talkbase and were answered by talkbase Attendant. Calls cannot be made to that SIP user directly but only to the main number. The SIP user was configured without any Application Sequences.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by

DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and talkbase did not include use of any specific encryption features to Session Manager as requested by FROX AG. The SIP trunk between talkbase Server and Session Manager was configured using TCP.

2.1. Interoperability Compliance Testing

Interoperability compliance testing included feature and serviceability testing. The testing for talkbase Attendant focused on the following functionality:

- login/logout attendants.
- make/receive basic calls.
- make/receive PSTN calls.
- hold/transfer/forward calls.

The serviceability testing focused on verifying the ability of the talkbase solution to recover from network outage of the talkbase server, talkbase Attendant, and Session Manager. Recovery from talkbase server reboot was also verified.

2.2. Test Results

All test cases were executed and verified. All test cases passed successfully. The following observations were noted during interoperability testing.

- G.711MU and G.711A codecs were verified during testing. Other codecs can be configured through talkbase and their dialogic licensing, but were not tested.
- Environments using H.323 endpoints are not supported due to issues with one way audio during calls with talkbase Attendant. Interoperability with H.323 endpoints in the Avaya environment are not supported by talkbase.
- Setting call forwarding from talkbase Attendant is not supported.
- The caller display of an incoming PSTN call made to an Avaya SIP endpoint that was blind transferred to talkbase Attendant did not show the PSTN number but the (transferring) Avaya SIP endpoint number. This is a known problem with talkbase and may be fixed in a future release.
- Since SIP users are administered without application sequences, features requiring feature access codes activation are not supported.

2.3. Support

Technical support for Frox AG talkbase can be found as follows.

- **Phone:** +41 55 254 12 54/89
- **Email:** info@talkbase.com
- **Web:** <https://talkbase.com/en/>

3. Reference Configuration

Figure 1 shows the configuration used during compliance testing. The talkbase server is placed on the Avaya telephony LAN. Session Manager provides the talkbase SIP connection to Communication Manager. talkbase Attendant is capable of logging into an Avaya SIP endpoint and receiving calls via a web page on a client PC.

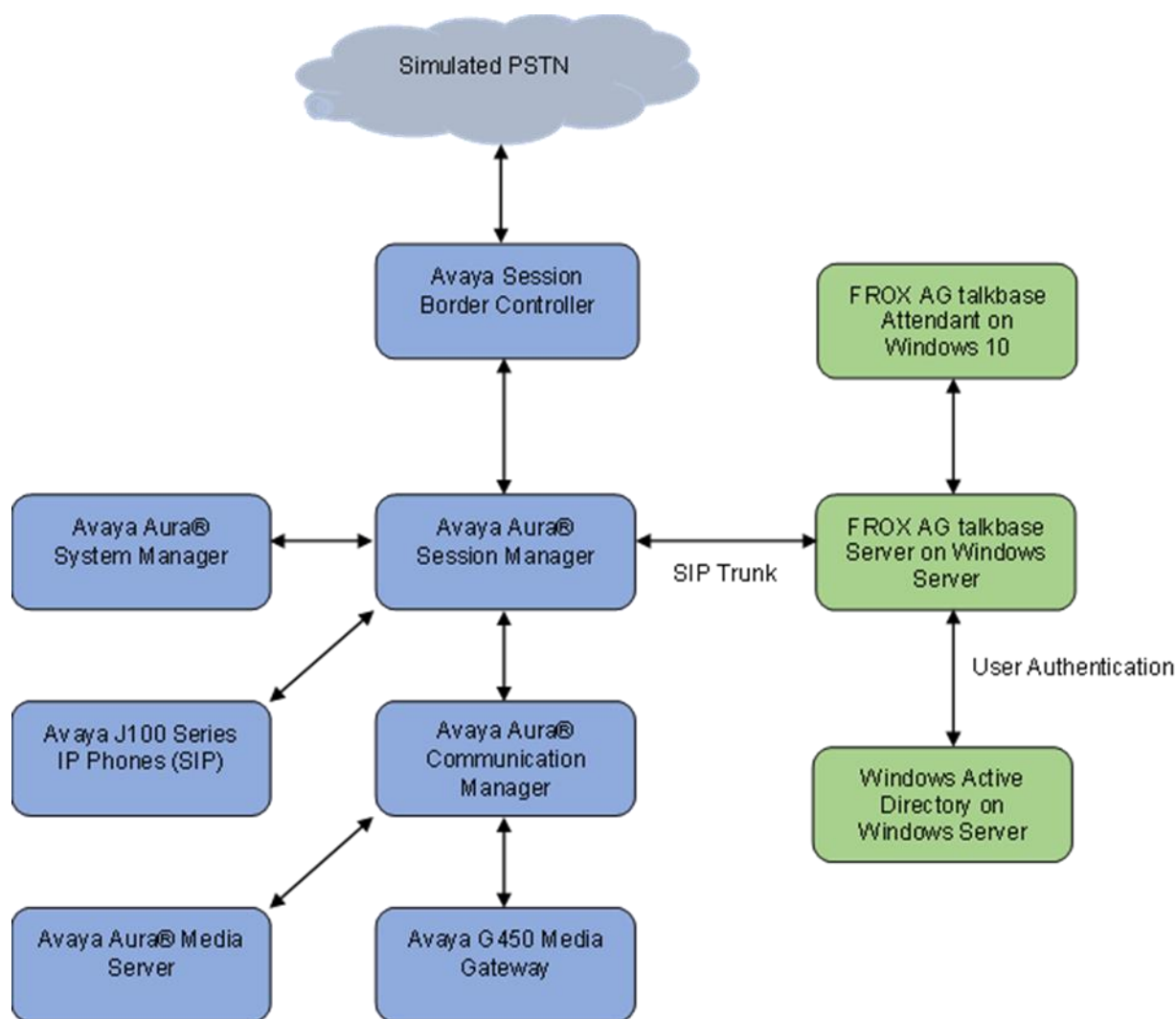


Figure 1: Network Solution of FROX AG talkbase and Avaya Aura® Communication Manager and Avaya Aura® Session Manager

4. Equipment and Software Validated

The following equipment and software versions are used for the sample configuration:

Equipment / Software	Release / Version
Avaya Aura® System Manager running on Virtual Machine	10.1.0.2 Service Pack 2 10.1.0.2.0715160
Avaya Aura® Session Manager running on Virtual Machine	10.1.0.2 Service Pack 10.1.0.02.1010215
Avaya Aura® Communication Manager running on Virtual Machine	10.1.0.2-SP2 01.0.974.0-27607
Avaya Session Border Controller for Enterprise running on Virtual Machine	10.1.1.0-35-21872
Avaya Aura® Media Server running on Virtual Machine	10.1.0.101
Avaya J139/J179/J189 SIP Deskphone	4.0.13.0.6
Avaya G450 Media Gateway	42.7.0
FROX AG talkbase server running on Windows Server 2019	22.02-4
talkbase Attendant on Windows 10 PC	22.02-4 on Chrome 106.0.5249.119 (Official Build) (64-bit)

5. Configure Avaya Aura® Communication Manager

The configuration and verification operations illustrated in this section are all performed using Communication Manager System Administration Terminal (SAT). The information provided in this section describes the configuration of Communication Manager for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**. The procedures include the following areas:

- Verify System Parameters Customer Options
- Verify System Features
- Configure SIP Trunk
- Administer Call Routing

Note: The configuration of the simulated PSTN is outside the scope of these Application Notes.

5.1. Verify System Parameters Customer Options

The license file installed on the system controls these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative. Use the **display system-parameters customer-options** command to determine these values. On **Page 2**, verify that the **Maximum Administered SIP Trunks** have sufficient capacity. Each call uses a minimum of one SIP trunk.

display system-parameters customer-options		Page	2 of 12
OPTIONAL FEATURES			
IP PORT CAPACITIES		USED	
Maximum Administered H.323 Trunks:	12000	0	
Maximum Concurrently Registered IP Stations:	2400	4	
Maximum Administered Remote Office Trunks:	12000	0	
Max Concurrently Registered Remote Office Stations:	2400	0	
Maximum Concurrently Registered IP eCons:	128	0	
Max Concur Reg Unauthenticated H.323 Stations:	100	0	
Maximum Video Capable Stations:	36000	1	
Maximum Video Capable IP Softphones:	150	21	
Maximum Administered SIP Trunks:	12000	20	
Max Administered Ad-hoc Video Conferencing Ports:	12000	0	
Max Number of DS1 Boards with Echo Cancellation:	688	0	

5.2. Verify System Features

For compliance testing, **Trunk-to Trunk Transfer** is set to **all** on **Page 1** of the **system-parameters features** page. This is a system wide setting that allows calls to be routed from one trunk to another and is usually turned off to help prevent toll fraud. An alternative to enabling this feature on a system wide basis is to control it using COR (Class of Restriction). See **Section 10** for supporting documentation.

```
display system-parameters features                                     Page 1 of 19
      FEATURE-RELATED SYSTEM PARAMETERS
      Self Station Display Enabled? n
      Trunk-to-Trunk Transfer: all
      Automatic Callback with Called Party Queuing? n
      Automatic Callback - No Answer Timeout Interval (rings): 8
      Call Park Timeout Interval (minutes): 10
      Off-Premises Tone Detect Timeout Interval (seconds): 20
      AAR/ARS Dial Tone Required? y

      Music (or Silence) on Transferred Trunk Calls? no
      DID/Tie/ISDN/SIP Intercept Treatment: attendant
      Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
      Automatic Circuit Assurance (ACA) Enabled? n

      Abbreviated Dial Programming by Assigned Lists? n
      Auto Abbreviated/Delayed Transition Interval (rings): 2
      Protocol for Caller ID Analog Terminals: Bellcore
      Display Calling Number for Room to Room Caller ID Calls? n
```


5.3. Configure SIP Trunk

In the **Node Names IP** form, note the IP Address of Communication Manager (**procr**) and Session Manager (**sm10**). The host names will be used throughout the other configuration screens of Communication Manager and Session Manager. Type **display node-names ip** to see node names.

```
display node-names ip
```

IP NODE NAMES	
Name	IP Address
aes10	10.64.110.247
aes811	10.64.110.209
ams10	10.64.110.214
aura_cms18	10.64.110.20
cms19	10.64.110.225
default	0.0.0.0
procr	10.64.110.213
procr6	::
remotecms191	10.64.110.226
sm10	10.64.110.212

(10 of 10 administered node-names were displayed)
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name

In the **IP Network Region** form, the **Authoritative Domain** field is that configured on Session Manager. In this configuration, the domain name is **avaya.com**. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the G450 Media Gateway or Media Server. The **IP Network Region** form also specifies the **IP Codec Set** to be used. This codec set is used for calls routed over the SIP trunk to Session Manager as **ip-network region 1** is specified in the SIP signaling group.

```
display ip-network-region 1
```

Page 1 of 20

IP NETWORK REGION	
Region: 1	NR Group: 1
Location: 1	Authoritative Domain: avaya.com
Name: Main	Stub Network Region: n
MEDIA PARAMETERS	Intra-region IP-IP Direct Audio: yes
Codec Set: 1	Inter-region IP-IP Direct Audio: yes
UDP Port Min: 2048	IP Audio Hairpinning? n
UDP Port Max: 3329	
DIFFSERV/TOS PARAMETERS	
Call Control PHB Value: 46	
Audio PHB Value: 46	
Video PHB Value: 26	
802.1P/Q PARAMETERS	
Call Control 802.1p Priority: 6	
Audio 802.1p Priority: 6	
Video 802.1p Priority: 5	
H.323 IP ENDPOINTS	AUDIO RESOURCE RESERVATION PARAMETERS
H.323 Link Bounce Recovery? y	RSVP Enabled? n
Idle Traffic Interval (sec): 20	
Keep-Alive Interval (sec): 5	
Keep-Alive Count: 5	

In the **IP Codec Set** form, select the audio codecs supported for calls routed over the SIP trunk to talkbase. **IP codec set 1** is specified in **IP Network Region 1** previously discussed. Multiple codecs may be specified in the **IP Codec Set** form in order of preference. The example below includes **G.711A** (a-law) and **G.711MU** (mu-law) which are both supported by talkbase.

Media Encryption is used on the Avaya sets where possible. **None** is also present to facilitate any extension not capable of handling encryption. Calls to talkbase Attendant did not use Media Encryption.

display ip-codec-set 1				Page 1 of 2
IP MEDIA PARAMETERS				
Codec Set: 1				
Audio	Silence	Frames	Packet	
Codec	Suppression	Per Pkt	Size (ms)	
1: G.711MU	n	2	20	
2: G.711A	n	2	20	
3:				
4:				
5:				
6:				
7:				
Media Encryption		Encrypted SRTCP: best-effort		
1: 1-srtp-aescm128-hmac80				
2: 10-srtp-aescm256-hmac80				
3: none				
4:				
5:				

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the Signaling Group form as follows:

- Set the **Group Type** field to **sip**.
- Set the **Transport Method** to the desired transport method. **tls** (Transport Layer Security) should be used for DevConnect testing.
- The **Peer Detection Enabled** field should be set to **y** allowing Communication Manager to automatically detect if the peer server is a Session Manager.
- Set the **Near-end Node Name** to **procr**. This value is taken from the **IP Node Names** form shown above.
- Set the **Far-end Node Name** to the node name defined for the Session Manager (node name **sm10**).
- Ensure that the TLS port value of **5070** is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- In the **Far-end Network Region** field, enter the IP Network Region (e.g., **1**) configured above. This field logically establishes the **far-end** for calls using this signaling group as network region **1**.
- The **Far-end Domain** field is set to the domain name specified in the IP Network Region (e.g., **avaya.com**).
- The **DTMF over IP** field should remain set to the default value of **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- The **Direct IP-IP Audio Connections** field is set to **y**.
- **Initial IP-IP Direct Media** is also set to **y** to allow the RTP to setup directly between talkbase and the caller.
- The default values for the other fields may be used.

Note: These are settings for compliance testing, however, this trunk may be setup differently on each customer site depending on the customer's requirements for SIP routing.

change signaling-group 12		Page 1 of 3
SIGNALING GROUP		
Group Number: 12	Group Type: sip	
IMS Enabled? n	Transport Method: tls	
Q-SIP? n		
IP Video? n	Enforce SIPS URI for SRTP? n	
Peer Detection Enabled? y	Peer Server: SM	Clustered? n
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y		
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n		
Alert Incoming SIP Crisis Calls? n		
Near-end Node Name: procr	Far-end Node Name: sm10	
Near-end Listen Port: 5070	Far-end Listen Port: 5070	
	Far-end Network Region: 1	
Far-end Domain: avaya.com		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? y	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Initial IP-IP Direct Media? y	
	Alternate Route Timer(sec): 6	

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to and from talkbase. Enter a descriptive name in the **Group Name** field. Set the **Group Type** field to **sip**. Enter a **TAC** code compatible with the Communication Manager dial plan. Set the **Service Type** field to **tie**. Specify the signaling group associated with this trunk group in the **Signaling Group** field and specify the **Number of Members** supported by this SIP trunk group. Accept the default values for the remaining fields.

change trunk-group 12		Page 1 of 5	
TRUNK GROUP			
Group Number: 12	Group Type: sip	CDR Reports: y	
Group Name: talkbase	COR: 1	TN: 1	TAC: 112
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n			
Queue Length: 0			
Service Type: tie	Auth Code? n		
	Member Assignment Method: auto		
	Signaling Group: 12		
	Number of Members: 10		

On **Page 3** of the trunk-group form the **Numbering Format** is set to **private**. The rest of the fields are set as shown.

change trunk-group 12	Page 3 of 5
TRUNK FEATURES	
ACA Assignment? n	Measured: none
	Maintenance Tests? y
Suppress # Outpulsing? n	Numbering Format: private
	UI Treatment: shared
	Maximum Size of UI Contents: 128
	Replace Restricted Numbers? n
	Replace Unavailable Numbers? n
	Modify Tandem Calling Number: no
Send UCID? n	
Show ANSWERED BY on Display? y	
DSN Term? n	

Settings on **Page 5** are as follows.

change trunk-group 12	Page 5 of 5
PROTOCOL VARIATIONS	
	Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n	
Send Transferring Party Information? y	
Network Call Redirection? y	
Build Refer-To URI of REFER From Contact For NCR? n	
Send Diversion Header? n	
Support Request History? y	
Telephone Event Payload Type: 101	
Convert 180 to 183 for Early Media? n	
Always Use re-INVITE for Display Updates? n	
Resend Display UPDATE Once on Receipt of 481 Response? n	
Identity for Calling Party Display: P-Asserted-Identity	
Block Sending Calling Party Location in INVITE? n	
Accept Redirect to Blank User Destination? n	
Enable Q-SIP? n	
Interworking of ISDN Clearing with In-Band Tones: keep-channel-active	
Request URI Contents: may-have-extra-digits	

5.4. Administer Call Routing

A range of numbers is assigned to the Attendant as main numbers where Communication Manager users can dial to or the PSTN can call. For compliance testing, the range of numbers made available is 70110 to 70119. 70115 and 70116 are used for talkbase main numbers during testing. The configuration must route calls 7011x to the SIP trunk created in **Section 5.3** and to do that a route pattern must be created before the routing can be set. This route pattern references the SIP trunk and the routing then references the route pattern.

5.4.1. Configure Route Pattern

Use the **change route-pattern 12** command to add the SIP trunk group to the route pattern that Automatic Alternate Routing configured in **Section 5.4.3** selects. In this configuration, Route Pattern Number **12** is used to route calls to trunk group **12** configured in **Section 5.3**. The **Numbering Format** is set to **lev0-pvt**.

change route-pattern 12												Page 1 of 3	
Pattern Number: 12												Pattern Name: Talkbase	
SCCAN? n		Secure SIP? n		Used for SIP stations? n									
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted	DCS/ IXC					
No			Mrk	Lmt	List	Del	Digits	QSIG					
							Dgts	Intw					
1:	12	0						n	user				
2:								n	user				
3:								n	user				
4:								n	user				
5:								n	user				
6:								n	user				
BCC VALUE		TSC	CA-TSC		ITC		BCIE	Service/Feature	PARM	Sub	Numbering	LAR	
0 1 2 M 4 W			Request							Dgts	Format		
1:	y	y	y	y	y	n	n	rest			lev0-pvt	none	
2:	y	y	y	y	y	n	n	rest				none	
3:	y	y	y	y	y	n	n	rest				none	
4:	y	y	y	y	y	n	n	rest				none	
5:	y	y	y	y	y	n	n	rest				none	
6:	y	y	y	y	y	n	n	rest				none	

5.4.2. Configure Uniform Dialplan

For compliance testing, all calls to talkbase Attendant are calls that began with **7011** and these are to be sent across the SIP trunk to Session Manager and then to talkbase. To achieve this routing, Automatic Alternate Routing (AAR) is used to route the calls. The dial plan and aar routing analysis configuration will allow this routing.

Use the **change uniform-dialplan** command to configure the routing of the dialed digits. In the example below, calls to **7011x** will use Automatic Alternate Routing (AAR). No further digits are deleted or inserted. Calls are sent to **aar** for further processing.

change uniform-dialplan 7					Page 1 of 2	
UNIFORM DIAL PLAN TABLE					Percent Full: 0	
Matching Pattern	Len	Del	Insert Digits	Net Conv	Node Num	
7011	5	0		aar	n	
7204	5	0		aar	n	
7205	5	0		aar	n	
76	5	0		aar	n	
					n	
					n	
					n	

5.4.3. Configure Automatic Alternate Routing

Use the **change aar analysis** command to further configure the routing. Calls to the Attendant are achieved by dialing **70115** or **70116** and are matched with the **Dialed String** entry shown below. Calls are sent to **Route Pattern 12**, configured in **Section 5.4.1**.

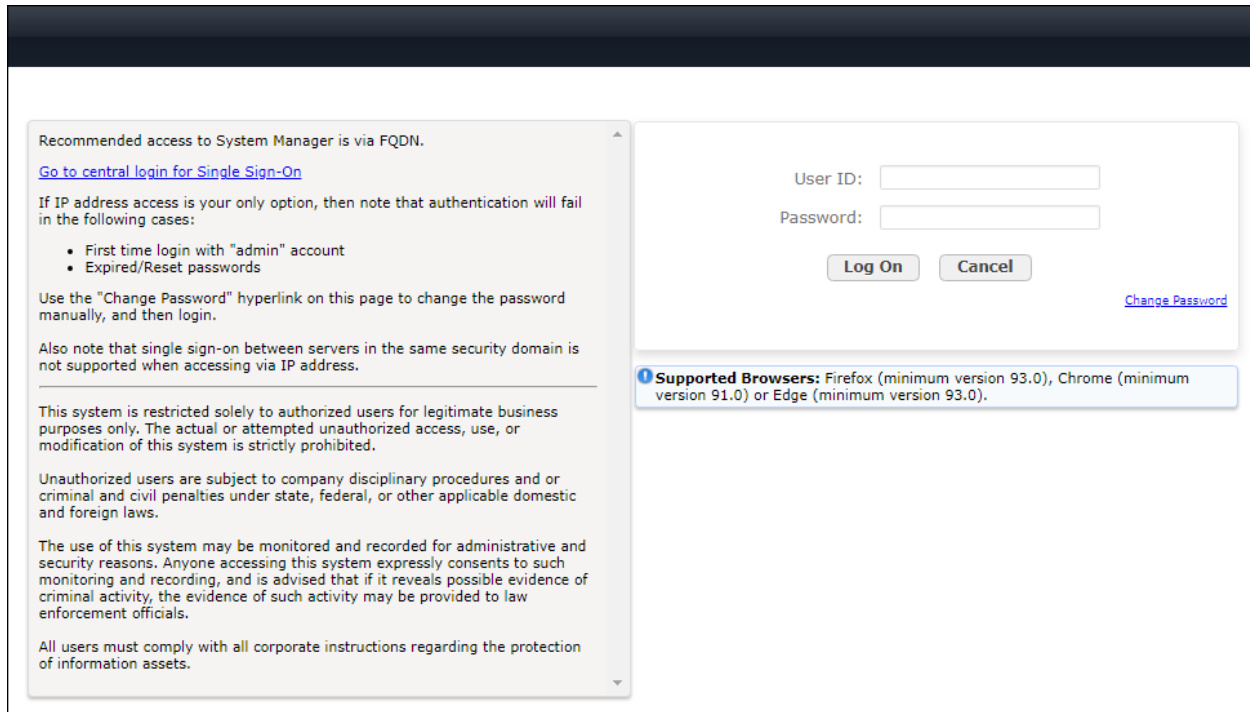
change aar analysis 7					Page 1 of 2	
AAR DIGIT ANALYSIS TABLE					Percent Full: 0	
Location: all						
Dialed String	Total		Route	Call	Node	ANI
	Min	Max	Pattern	Type	Num	Reqd
70	5	5	1	lev0		n
7011	5	5	12	lev0		n
71	5	5	1	aar		n
						n
						n
						n
						n
						n

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. Session Manager is configured via System Manager. The procedures include the following areas:

- Domains and Locations
- Configure Communication Manager SIP Entity
- Configure Communication Manager Entity Link
- Configure talkbase SIP Entity
- Configure talkbase Entity Link
- Configure Routing Policy for talkbase
- Configure Dial Patterns
- Configure talkbase SIP User

To make changes on Session Manager, a web session is established to System Manager. Log into System Manager by opening a web browser and navigate to <https://<System Manager FQDN>/SMGR>. Enter the appropriate credentials for the **User ID** and **Password** and click on **Log On**.



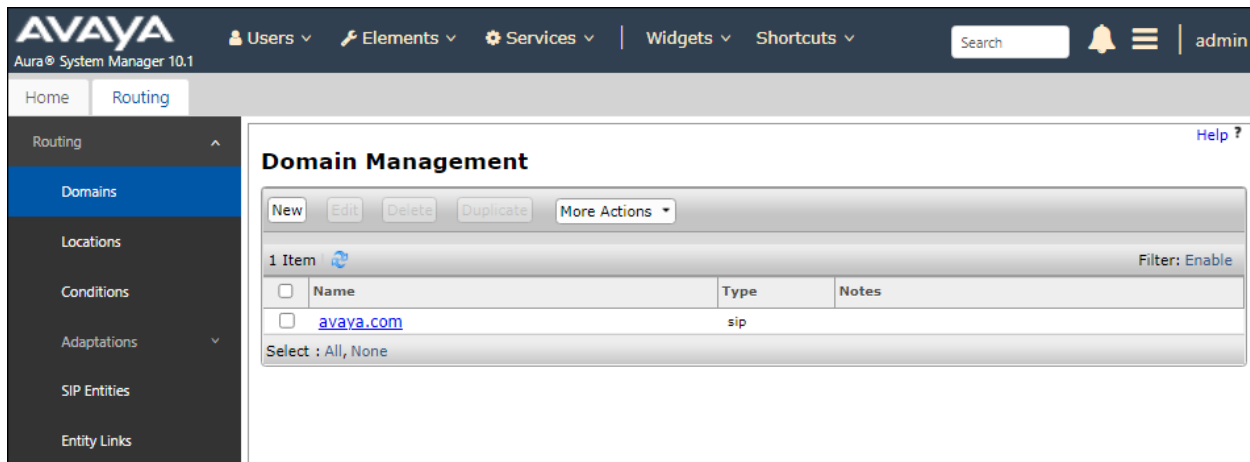
Once logged in navigate to **Elements** and click on **Routing** (not shown).

6.1. Domains and Locations

Note: It is assumed that a domain and a location have already been configured. An overview of the domain and location that used in compliance testing is provided.

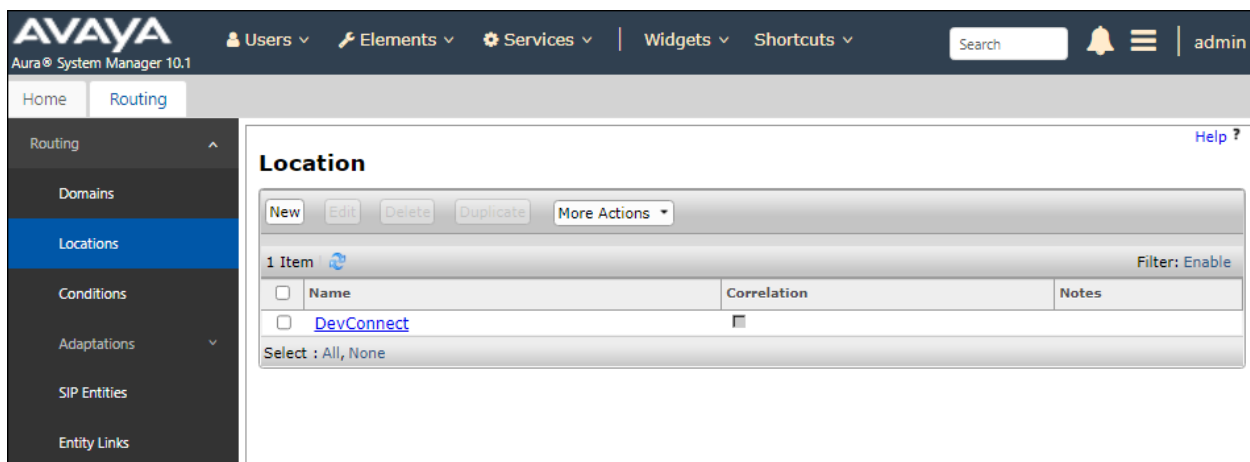
6.1.1. Display the Domain

Select **Domains** from the left window. This will display the domain configured on Session Manager. For compliance testing this domain is **avaya.com** as shown below. If a domain is not already in place, click New to add a new domain.



6.1.2. Display the Location

Select **Locations** from the left window. this will display the location configured on Session Manager. For compliance testing, this location is **DevConnect** as shown below. If a location is not already in place, click **New** to add a new location.



6.2. Configure Communication Manager SIP Entity

A SIP entity for Communication Manager should already be configured in the Avaya environment. SIP entity creation for Communication Manager is shown here for reference.

Click on **SIP Entities** in the left column and select **New** in the right window (not shown). Enter a suitable **Name** for the Communication Manager SIP Entity and the **IP Address** of Communication Manager. Select **CM** for **Type**. Enter the correct **Time Zone**. An adaptation may be used if attendants are required to forward calls to their assigned main number. Reference [6] for details. If needed, click the **Add** button and select the Adaptation created for talkbase from **Section 6.9.2**. In this case, the adaptation **cm10 remove 789** is used. Accept the default values for the remaining fields. Click **Commit**.

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Home Routing

Routing ▾

- Domains
- Locations
- Conditions
- Adaptations ▾
- SIP Entities**
- Entity Links
- Time Ranges
- Routing Policies
- Dial Patterns ▾
- Regular Expressions
- Defaults

SIP Entity Details Commit Cancel Help ?

General

* Name:

* FQDN or IP Address:

Type:

Notes:

Location:

Time Zone:

* SIP Timer B/F (in seconds):

Minimum TLS Version:

Credential name:

Securable: ☐

Call Detail Recording:

Adaptations

<input type="checkbox"/>	Order	Name	Module Name	State	Type	Notes
<input type="checkbox"/>	1	cm10 remove 789	DigitConversionAdapter	enabled	digit	

Select : All, None

6.3. Configure Communication Manager SIP Entity Link

The Entity Link can be added from the Communication Manager SIP Entity page from **Section 6.2**. Scroll down to expose the **Entity Links** Section. Click on the **Add** button. Select the **Session Manager** SIP Entity (e.g., **sm10**) for **SIP Entity 1** and the newly created **Communication Manager** SIP Entity (e.g., **cm10_Talkbase**) for **SIP Entity 2**. Ensure that **TLS** is selected for the **Protocol** and that **Port 5070** to match the port values used in **Section 5.3**. Ensure **Trusted** is selected for **Connection Policy**. Click on **Commit** to save the new Entity Link.

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Home Routing

Routing Domains Locations Conditions Adaptations ▾ SIP Entities Entity Links Time Ranges Routing Policies Dial Patterns ▾ Regular Expressions <

Backup Session Manager Bandwidth Association: ▾

Entity Links

Override Port & Transport with DNS SRV: ☐

Add Remove

1 Item Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Po
<input type="checkbox"/>	* sm10_cm10_Talkbase_5	sm10	TLS ▾	* 5070	cm10_Talkbase	*

Select : All, None

SIP Responses to an OPTIONS Request

Add Remove

0 Items Filter: Enable

<input type="checkbox"/>	Response Code & Reason Phrase	Mark Entity Up/Down	Notes
--------------------------	-------------------------------	---------------------	-------

Commit Cancel

6.4. Configure talkbase SIP Entity

Click on **SIP Entities** in the left column and select **New** in the right window (not shown). Enter a suitable **Name** for the new talkbase SIP Entity and the **IP Address** of the talkbase server. Select **SIP Trunk** for **Type**. Enter the correct **Time Zone**. An adaptation can be used if attendants are required to forward calls to their assigned main number. Reference [6] for details. Click the **Add** button and select the Adaptation created for talkbase from **Section 6.9.1**. In this case, the adaptation **Talkbase prepend 789** is used. Accept the default values for the remaining fields. Click **Commit**.

AVAYA Aura® System Manager 10.1

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Home Routing

Routing Domains Locations Conditions Adaptations ▾ **SIP Entities** Entity Links Time Ranges Routing Policies Dial Patterns ▾ Regular Expressions Defaults

SIP Entity Details Commit Cancel Help ?

General

* Name: TalkbaseServer

* FQDN or IP Address: 10.64.110.151

Type: SIP Trunk ▾

Notes:

Location: ▾

Time Zone: America/Denver ▾

* SIP Timer B/F (in seconds): 4

Minimum TLS Version: Use Global Setting ▾

Credential name:

Securable: ☐

Call Detail Recording: egress ▾

Adaptations

Add Remove

<input type="checkbox"/>	Order	Name	Module Name	State	Type	Notes
<input type="checkbox"/>	1	Talkbase prepend 789 ▾	DigitConversionAdapter	enabled	digit	

Select : All, None

6.5. Configure talkbase SIP Entity Link

The Entity Link can be added from the talkbase SIP Entity page from **Section 6.4**. Scroll down to expose the **Entity Links** Section. Click on the **Add** button. Select the **Session Manager** SIP Entity (e.g., **sm10**) for **SIP Entity 1** and the newly created talkbase SIP Entity (e.g., **TalkbaseServer**) for **SIP Entity 2**. Ensure that **TCP** is selected for the **Protocol** and that **Port 5060** is used. Ensure **Trusted** is selected for **Connection Policy**. Click on **Commit** to save the new Entity Link.

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Aura® System Manager 10.1

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SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns ▾

Entity Links

Add Remove

1 Item Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
<input type="checkbox"/>	* sm10_TalkbaseServer_5	sm10	TCP	* 5060	TalkbaseServer	* 5060	trusted	<input type="checkbox"/>

Select : All, None

Failover Ports

TCP Failover port:

TLS Failover port:

Listen Ports

Add Remove

0 Items Filter: Enable

<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Endpoint	Notes
--------------------------	--------------	----------	----------------	----------	-------

6.6. Configure Routing Policy for talkbase

Click on **Routing Policies** in the left window and select **New** in the main window. Enter a suitable **Name** for the Routing Policy and click the **Select** button under **SIP Entity as Destination**. Select the SIP entity created in **Section 6.2** (e.g., **TalkbaseServer**) and click **Select**.

The selected destination is now shown. Click **Commit** to save the Routing Policy.

The screenshot displays the Avaya Aura System Manager 10.1 interface. The left sidebar shows the navigation menu with 'Routing Policies' selected. The main content area is titled 'Routing Policy Details' and includes a 'Commit' button and a 'Cancel' button. The 'General' section contains fields for 'Name' (set to 'To Talkbase'), 'Disabled' (checkbox), 'Retries' (set to 0), and 'Notes'. The 'SIP Entity as Destination' section features a 'Select' button and a table with columns: Name, FQDN or IP Address, Type, and Notes. The table lists 'TalkbaseServer' with FQDN '10.64.110.151' and Type 'SIP Trunk'. The 'Time of Day' section includes 'Add', 'Remove', and 'View Gaps/Overlaps' buttons, a 'Filter: Enable' link, and a table with columns: Ranking, Name, Mon, Tue, Wed, Thu, Fri, Sat, Sun, Start Time, End Time, and Notes. The table shows one item with Ranking 0, Name 24/7, and Start/End times 00:00 to 23:59. A 'Select : All, None' link is at the bottom.

Name	FQDN or IP Address	Type	Notes
TalkbaseServer	10.64.110.151	SIP Trunk	

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	☑	☑	☑	☑	☑	☑	☑	00:00	23:59	Time Range 24/7

A dial pattern must be created to route talkbase main numbers to the talkbase SIP entity. Select **Dial Patterns** in the left window and select **New** in the main window. Enter the required digits for **Pattern**, e.g., **7011**, and the appropriate **Min** and **Max** digits. Click on **Add** under **Originating Locations and Routing Policies** and select the appropriate **Originating Location Name** and the Routing Policy from **Section 6.6** (e.g., **To Talkbase**) for **Routing Policy Name**. Click the **Select** button. Click **Commit** to finish adding the Dial Pattern. The configuration below will route calls to extensions **7011x** to talkbase.

RH; Reviewed:
SPOC 12/7/2022

6.8. Configure talkbase SIP User

A SIP user must be configured for talkbase Attendant. In Session Manager, select **Users → User Management → Manage Users** to display the **User Management** screen (not shown). Click + **New** to add a user. Under the **Identity** tab, enter an appropriate **Last Name** and **First Name**. Enter <extension>@<sip domain> of the user (e.g., **70111@avaya.com**).

Home / Users / Manage Users Help ?

User Profile | Add

[Commit & Continue](#) [Commit](#) [Cancel](#)

Identity | Communication Profile | Membership | Contacts

Basic Info

Address

LocalizedName

User Provisioning Rule:

* Last Name: Last Name (in Latin alphabet characters):

* First Name: First Name (in Latin alphabet characters):

* Login Name: Middle Name:

Description: Email Address:

Password: User Type:

Confirm Password: Localized Display Name:

Endpoint Display Name: Title Of User:

Language Preference: Time Zone:

Employee ID: Department:

Select the **Communication Profile** tab. Select **Communication Address** in the left list and click + **New**. Select **Avaya SIP** from the drop-down list for **Type**. Enter the extension number (e.g.,**70111**) for **Fully Qualified Address**. Enter the domain from **Section 6.1.1** (e.g., **avaya.com**). Click **OK**.

The screenshot displays the 'User Profile | Add' interface. The 'Communication Profile' tab is selected. In the left sidebar, 'Communication Address' is highlighted under the 'PROFILES' section. A modal dialog titled 'Communication Address Add/Edit' is open in the center. The dialog contains two required fields: '* Type:' with a dropdown menu showing 'Avaya SIP', and '* Fully Qualified Address:' with a text input containing '70111' and a domain dropdown showing '@ avaya.com'. At the bottom of the dialog are 'Cancel' and 'OK' buttons. The background interface shows a table with columns 'Type', 'Handle', and 'Domain', and a list of profiles including 'Session Manager Profile' and 'CM Endpoint Profile'.

Select **Communication Profile Password** on the left and in the **Comm-Profile Password** and **Re-enter Comm-Profile Password** fields, enter a password. This will be used to register the device. Click **OK**.

The screenshot shows a web application interface for adding a user profile. The main dialog is titled "User Profile | Add" and has tabs for "Identity", "Communication", and "PROFILES". The "Communication" tab is selected. A sub-dialog titled "Comm-Profile Password" is open in the foreground. This sub-dialog has two password input fields: "Comm-Profile Password :" and "* Re-enter Comm-Profile Password :". Both fields contain masked characters (dots). The second field has a green checkmark icon to its right, indicating the passwords match. Below the fields is a blue link labeled "Generate Comm-Profile Password". At the bottom of the sub-dialog are "Cancel" and "OK" buttons. The background interface includes a breadcrumb "Home / Users / Manage Users", a "Help" link, and buttons for "Commit & Continue", "Commit", and "Cancel".

Click on the **Session Manager Profile** slide button. For **Primary Session Manager**, select the values corresponding to the applicable Session Manager (e.g., **sm10**). Do not select any **Application Sequences**. Retain the default values in the remaining fields.

Home / Users / Manage Users Help ?

User Profile | Add

Commit & Continue Commit Cancel

- Identity
- Communication Profile
- Membership
- Contacts

Communication Profile Password

PROFILE SET : Primary

Communication Address

PROFILES

- Session Manager Profile ☒
- CM Endpoint Profile ☐

SIP Registration

* Primary Session Manager:

Secondary Session Manager:

Survivability Server:

Max. Simultaneous Devices:

Block New Registration When Maximum ☐

Application Sequences

Origination Sequence:

Click on the **CM Endpoint Profile**, slide button. Select the appropriate System (e.g., **cm10**), Endpoint for **Profile Type**, the Extension (e.g., **70111**). Click on **Commit** to save the new user.

Home / Users / Manage Users Help ?

User Profile | Add

Commit & Continue
Commit
Cancel

Identity

Communication Profile

Membership

Contacts

Communication Profile Password

PROFILE SET : Primary ▼

Communication Address

PROFILES

Session Manager Profile ☒

CM Endpoint Profile ☒

* System : cm10 ▼

Use Existing Endpoints : ☐

* Template : 9608SIP_DEFAULT_ 🔍

Security Code : Enter Security Code

Voice Mail Number :

Calculate Route Pattern : ☐

SIP URI : Select ▼

Override Endpoint Name and Localized Name : ☒

* Profile Type : Endpoint ▼

* Extension : 70111 🔍

* Set Type : 9608SIP

Port : IP 🔍

Preferred Handle : Select ▼

Sip Trunk : aar

Delete on Unassign from User or on Delete ☒

Allow H.323 and SIP Endpoint Dual Registration ☐

6.9. Configure for Attendant to Attendant Main Number Forwarding

This configuration is optional. In some situations, it may be desired for an attendant to forward a call back to their main number. Session Manager will route main number calls from attendants back to talkbase. talkbase cannot handle this case without additional configuration. In order to do this, call routing can be altered to have another call-id generated by routing these calls through Communication Manager. Refer to [6] for details. The below steps may be one such method to accomplish this.

6.9.1. Talkbase SIP entity Adaptation

Create a SIP entity adaptation to modify incoming main number call digits coming from talkbase Attendant in order to route them to Communication Manager instead of talkbase. Such an adaptation prepends digits to the main number called which will ultimately route the call to Communication Manager. The Adaptation **Talkbase prepend 789** below adds the digits **789** to the main numbers **70115** or **70116** dialed by attendants and can be assigned to the talkbase SIP entity as per **Section 6.4**.

The screenshot displays the Avaya Aura System Manager 10.1 interface. The left sidebar shows the navigation menu with 'Routing' selected. The main content area is titled 'Adaptation Details' and includes a 'Commit' button and a 'Cancel' button. The 'General' tab is active, showing the following configuration:

- Adaptation Name:** Talkbase prepend 789
- Notes:** (empty text area)
- Module Name:** DigitConversionAdapter (dropdown menu)
- Type:** digit
- State:** enabled (dropdown menu)
- Module Parameter Type:** Name-Value Parameter (dropdown menu)

Below these fields is a table for 'Module Parameters' with columns 'Name' and 'Value'. The table contains one entry: 'fromto' with a value of 'true'. There is an 'Add' button and a 'Remove' button above the table. Below the table is a 'Select : All, None' dropdown.

Below the 'Module Parameters' section is the 'Egress URI Parameters' section, which is currently empty.

Below the 'Egress URI Parameters' section is the 'Digit Conversion for Incoming Calls to SM' section. It includes an 'Add' button and a 'Remove' button. Below these buttons is a table with 9 columns: 'Matching Pattern', 'Min', 'Max', 'Phone Context', 'Delete Digits', 'Insert Digits', 'Address to modify', and 'Adaptation Data'. The table contains two rows of data:

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data
<input type="checkbox"/>	* 70115	* 5	* 5		* 0	789	destination	
<input type="checkbox"/>	* 70116	* 5	* 5		* 0	789	destination	

Below the table is a 'Select : All, None' dropdown.

6.9.2. Communication Manager SIP Entity Adaptation

Create a SIP entity adaptation to modify incoming main number calls from talkbase Attendant with modified digits to route them back to talkbase by removing the prepended digits. The Adaptation **cm10 remove 789** below removes the digits **789** prepended to the main numbers **70115** or **70116** dialed by attendants and can be assigned to the Communication Manager SIP entity as per **Section 6.2**.

The screenshot displays the Avaya Aura System Manager 10.1 interface. The left sidebar shows the navigation menu with 'SIP Entities' selected. The main content area is titled 'Adaptation Details' and includes a 'Commit' button and a 'Cancel' button. The 'General' tab is active, showing the following fields:

- Adaptation Name:** cm10 remove 789
- Notes:** (empty text area)
- Module Name:** DigitConversionAdapter (dropdown menu)
- Type:** digit (dropdown menu)
- State:** enabled (dropdown menu)
- Module Parameter Type:** Name-Value Parameter (dropdown menu)

Below these fields is a table for 'Module Parameters' with columns 'Name' and 'Value'. The table contains one row with 'fromto' in the 'Name' column and 'true' in the 'Value' column. There are 'Add' and 'Remove' buttons above the table, and a 'Select : All, None' option below it.

Below the table is the 'Egress URI Parameters' field, which is empty.

The 'Digit Conversion for Incoming Calls to SM' section shows a table with 0 items. The table has columns: Matching Pattern, Min, Max, Phone Context, Delete Digits, Insert Digits, Address to modify, Adaptation Data, and Notes. The 'Filter' is set to 'Enable'.

The 'Digit Conversion for Outgoing Calls from SM' section shows a table with 2 items. The table has columns: Matching Pattern, Min, Max, Phone Context, Delete Digits, Insert Digits, Address to modify, and Adaptation Data. The 'Filter' is set to 'Enable'.

Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data
* 78970115	8	8		3		destination	
* 78970116	8	8		3		destination	

6.9.3. Dial Pattern Route to talkbase SIP Entity

A dial pattern may be needed to ensure the modified queue call's destination routes to Communication Manager. A pattern was created using the method shown in **Section 6.7** to route destinations prepended with the digits **789** used in the adaptations created in **Sections 6.9.1 and 6.9.2** to the Communication Manager SIP entity **cm10_Talkbase**.

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Dial Patterns
Origination Dial...

Dial Pattern Details Commit Cancel [Help ?](#)

General

* Pattern: 789

* Min: 3

* Max: 36

Emergency Call: ☐

SIP Domain: -ALL- ▾

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item [Refresh](#) Filter: Enable

<input type="checkbox"/>	Originating Location Name ▴	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-		CM10-Talkbase	0	<input type="checkbox"/>	cm10_Talkbase	

Select : All, None

Denied Originating Locations

Add Remove

0 Items [Refresh](#)

<input type="checkbox"/>	Originating Location	Notes
--------------------------	----------------------	-------

7. Configure FROX AG talkbase

The configuration for talkbase to communicate with Session Manager is made on the talkbase server directly and a GUI using a web connection to the server. talkbase is dependent on having a Windows domain already in place with Windows Active Directory running. The talkbase server and talkbase Attendant client PC must be a part of this domain. talkbase users synchronize with Active Directory. The procedures include the following areas:

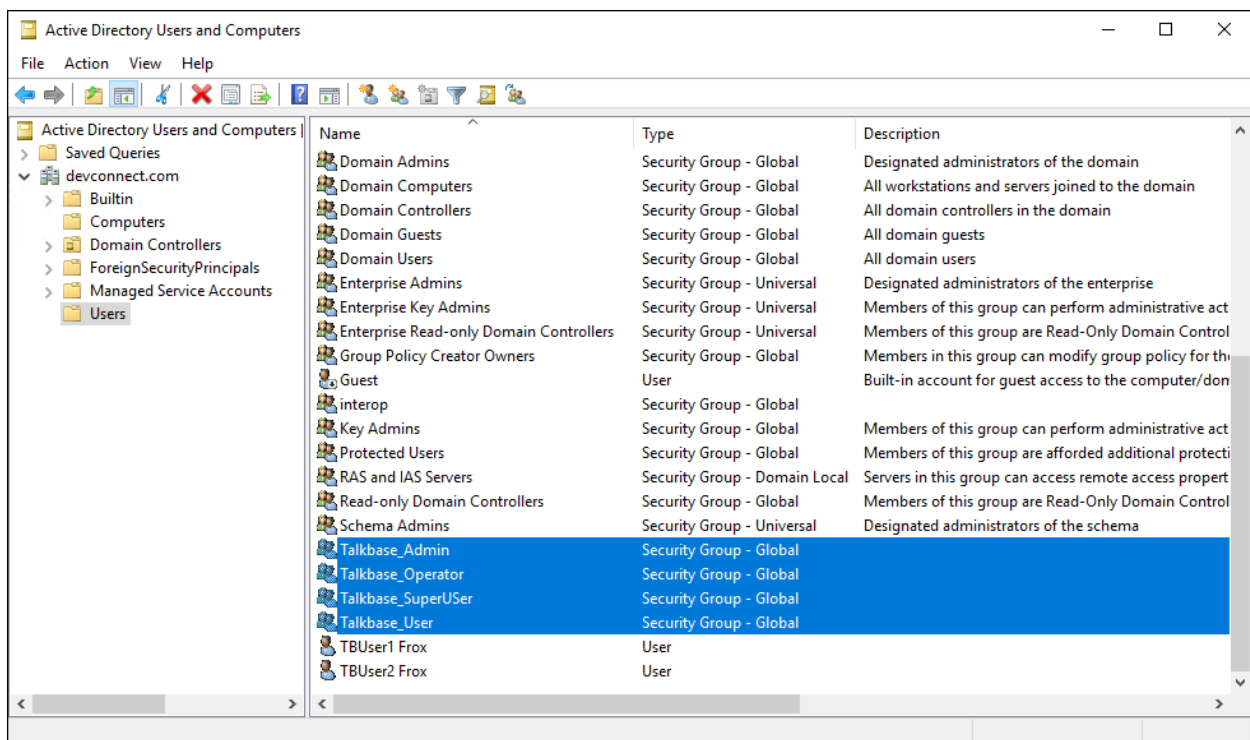
- Configure Active Directory for talkbase
- Configure the talkbase server
- Configure talkbase Attendant

7.1. Configure Active Directory for talkbase

The Active Directory must be configured for the following groups.

- **Talkbase_Admin**
- **Talkbase_Operator**
- **Talkbase_SuperUser**
- **Talkbase_User**

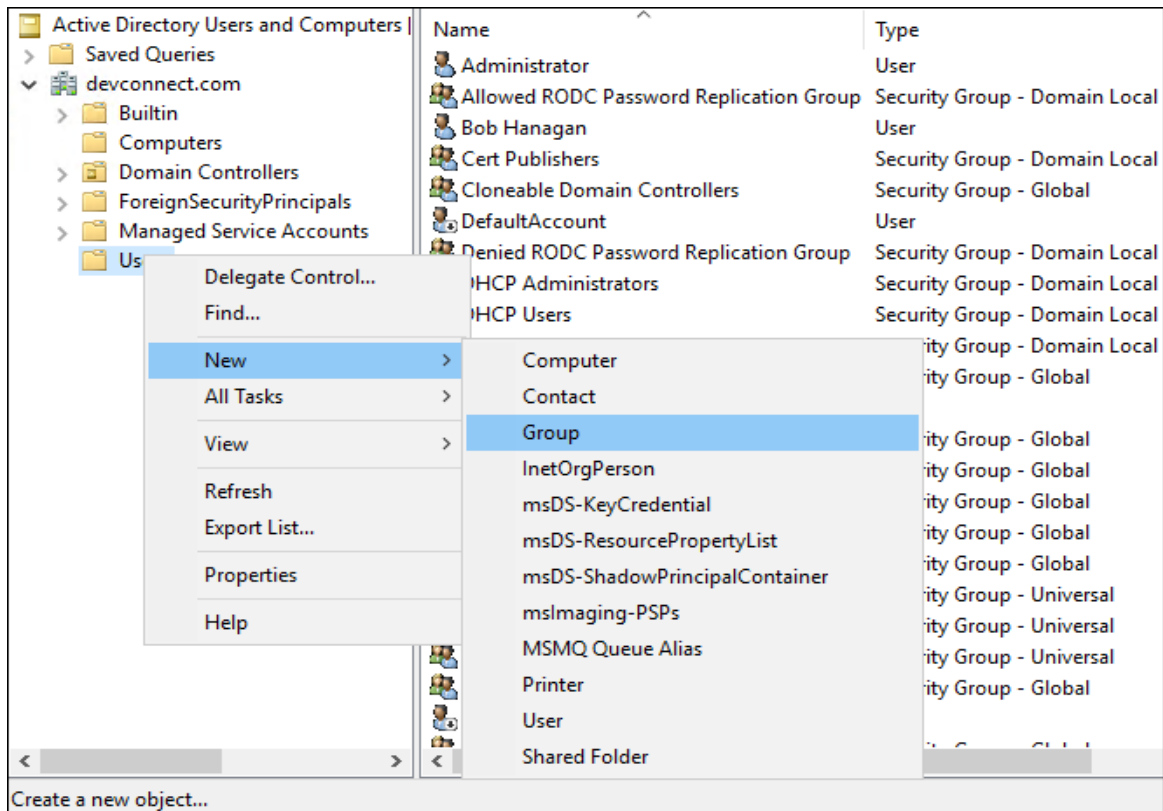
The various talkbase users that are added on Active Directory will be a part of one or more of these groups. The following shows these groups already configured.



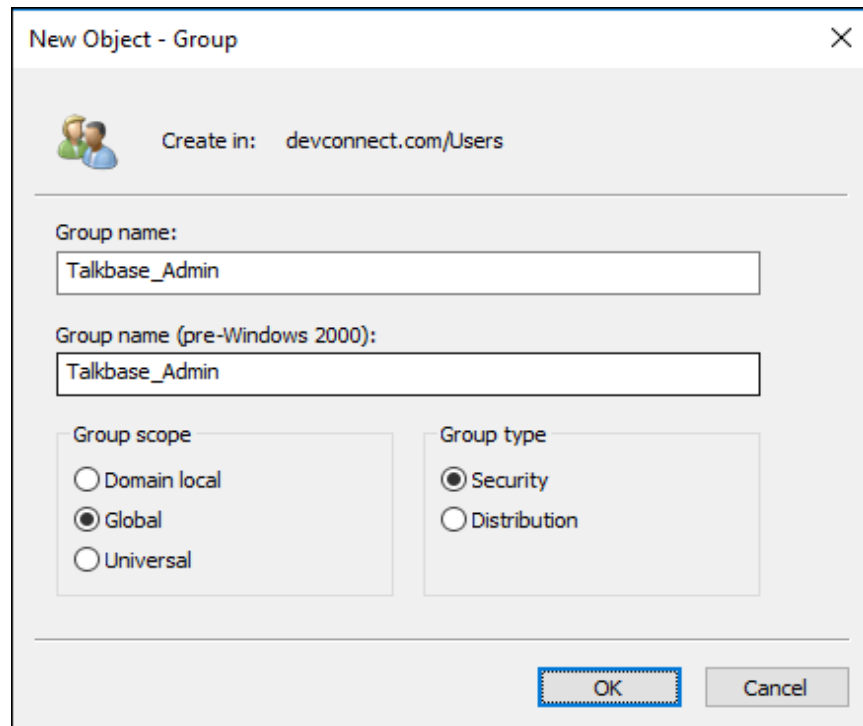
RH; Reviewed:
SPOC 12/7/2022

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Talkbase_CMSM10



Enter the **Group name** and ensure the **Group scope** and **Group type** are selected as shown below.



New Object - Group

Create in: devconnect.com/Users

Group name:
Talkbase_Admin

Group name (pre-Windows 2000):
Talkbase_Admin

Group scope

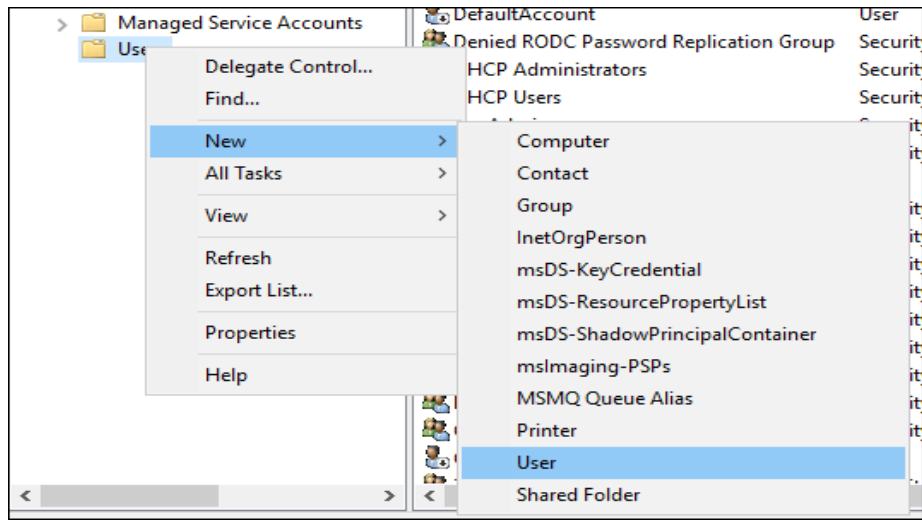
- ☐ Domain local
- ☒ Global
- ☐ Universal

Group type

- ☒ Security
- ☐ Distribution

OK Cancel

To add a new Active Directory user, right-click on **Users** and select **New** and then **User** as shown below.



The following shows information on a user called **Talkbase Attd 1** set up and used for testing. These are the users name details under the **General** tab.

A screenshot of the 'TBUser1 Frox Properties' dialog box, specifically the 'General' tab. The dialog box has a title bar with a question mark and a close button. Below the title bar are several tabs: 'Member Of', 'Dial-in', 'Environment', 'Sessions', 'Remote control', 'Remote Desktop Services Profile', 'COM+', 'General' (selected), 'Address', 'Account', 'Profile', 'Telephones', and 'Organization'. The 'General' tab displays a user icon and the name 'TBUser1 Frox'. Below this, there are input fields for 'First name' (containing 'TBUser1'), 'Initials' (empty), 'Last name' (containing 'Frox'), 'Display name' (containing 'Talkbase Attd 1'), 'Description' (empty), and 'Office' (empty). At the bottom, there are input fields for 'Telephone number', 'E-mail', and 'Web page', each with an 'Other...' button next to it. The 'OK' button is highlighted with a blue border.

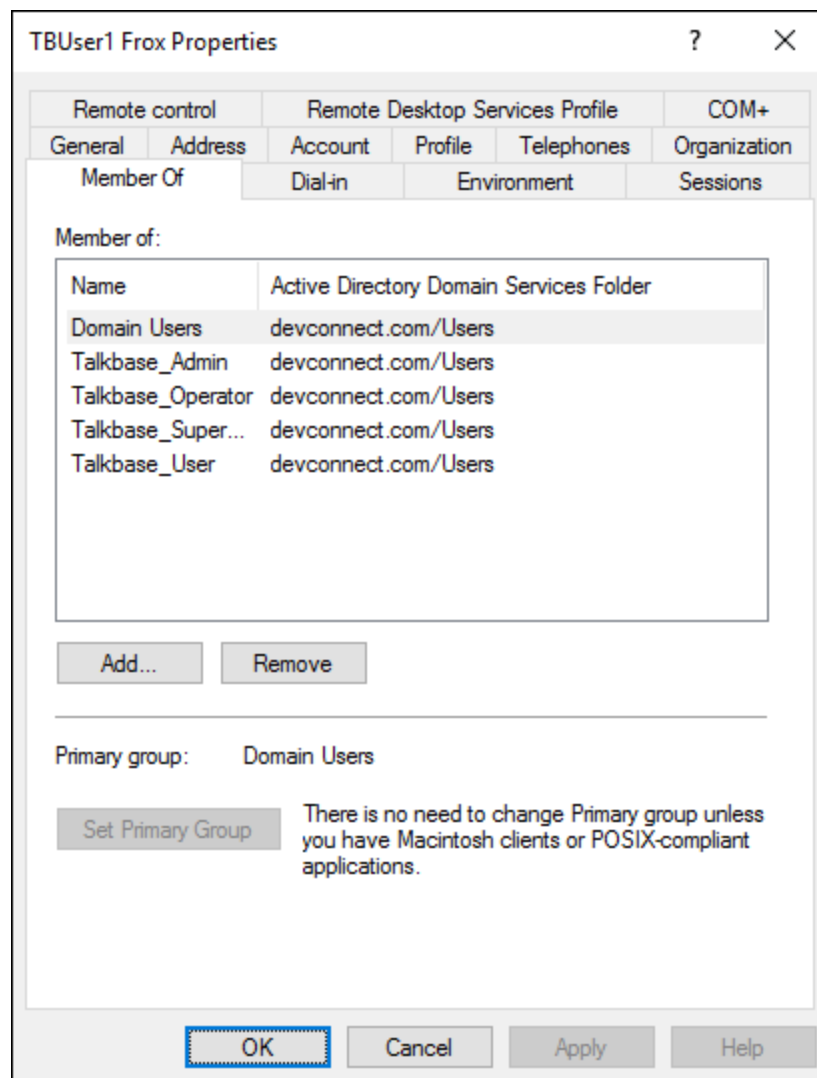
This is the account information under the **Account** tab.

The screenshot shows the 'TBUser1 Frox Properties' dialog box with the 'Account' tab selected. The dialog has a title bar with a question mark and a close button. Below the title bar is a tabbed interface with tabs for 'Member Of', 'Dial-in', 'Environment', 'Sessions', 'Remote control', 'Remote Desktop Services Profile', 'COM+', 'General', 'Address', 'Account' (selected), 'Profile', 'Telephones', and 'Organization'. The 'Account' tab contains the following fields and options:

- User logon name:** A text box containing 'TBUser1' and a dropdown menu showing '@devconnect.com'.
- User logon name (pre-Windows 2000):** Two text boxes, the first containing 'DEVCONNECT\' and the second containing 'TBUser1'.
- Logon Hours...** and **Log On To...** buttons.
- ☐ **Unlock account**
- Account options:** A list box containing four unchecked options:
 - ☐ User must change password at next logon
 - ☐ User cannot change password
 - ☐ Password never expires
 - ☐ Store password using reversible encryption
- Account expires:** A section with two radio buttons:
☒ **Never**
☐ **End of:** followed by a date picker showing 'Thursday , December 1, 2022'.

At the bottom of the dialog are four buttons: **OK**, **Cancel**, **Apply**, and **Help**.

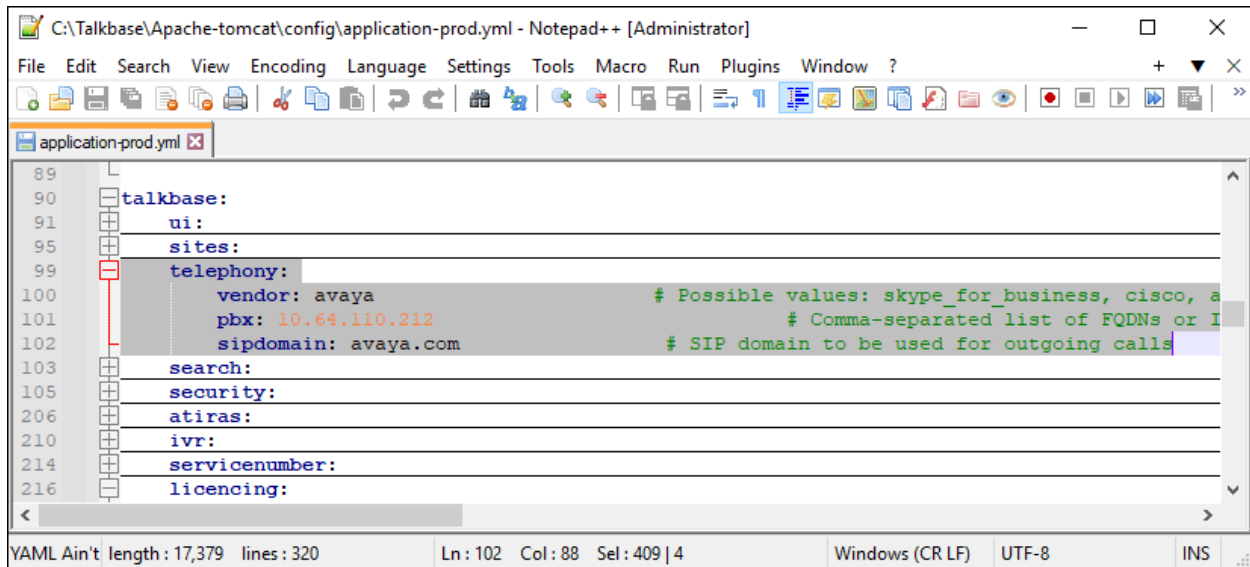
This particular user is set up to be a member of all four talkbase user groups, but this need not be the case on a customer site. Under the **Member Of** tab, the list of groups this user belongs to is displayed.



7.2. Configure the talkbase server

Configuration of the talkbase server is made by amending a file called **application-prod.yml**. The configuration file is created during the talkbase server installation and must be customized to specific installations. Essential modifications are detailed below. The file employed for compliance testing is included in **Appendix 1** for reference.

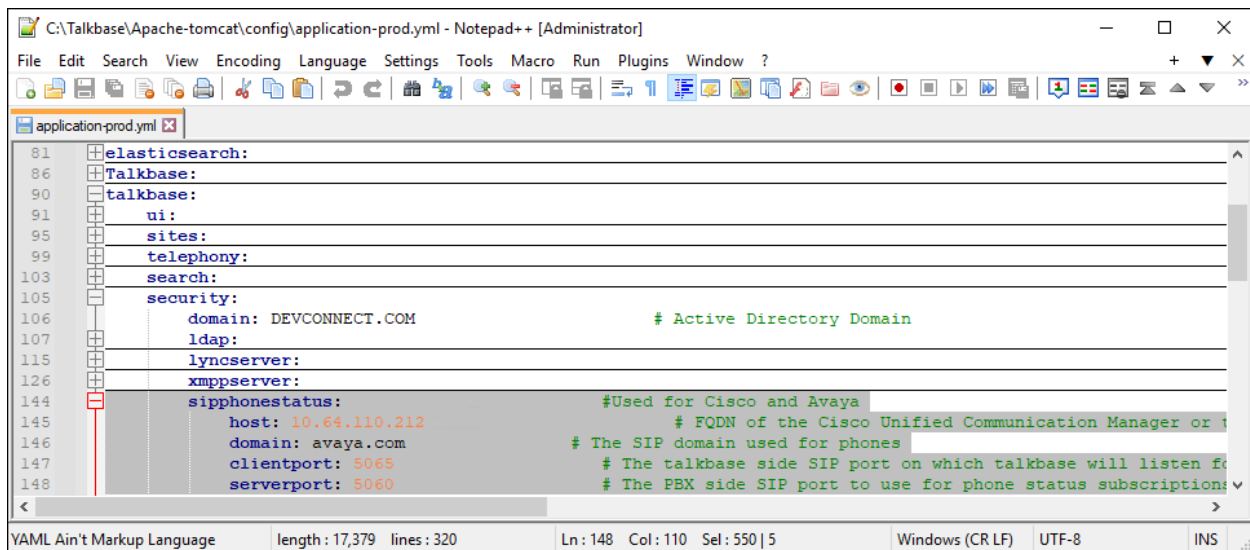
The talkbase server configuration file to modify is **C:\Talkbase\Apache-tomcat\config\application-prod.yml**. The file can be changed with a text editor. Under the **talkbase** block, verify the **Telephony** entries have the Session Manager IP Address (e.g., **10.64.110.212**) for **pbx** and the SIP Domain (e.g., **avaya.com**) for **sipdomain**.



```
89
90 talkbase:
91   ui:
95   sites:
99   telephony:
100     vendor: avaya # Possible values: skype_for_business, cisco, a
101     pbx: 10.64.110.212 # Comma-separated list of FQDNs or I
102     sipdomain: avaya.com # SIP domain to be used for outgoing calls
103   search:
105   security:
206   atiras:
210   ivr:
214   servicenumber:
216   licencing:
```

YAML Ain't length: 17,379 lines: 320 Ln: 102 Col: 88 Sel: 409 | 4 Windows (CR LF) UTF-8 INS

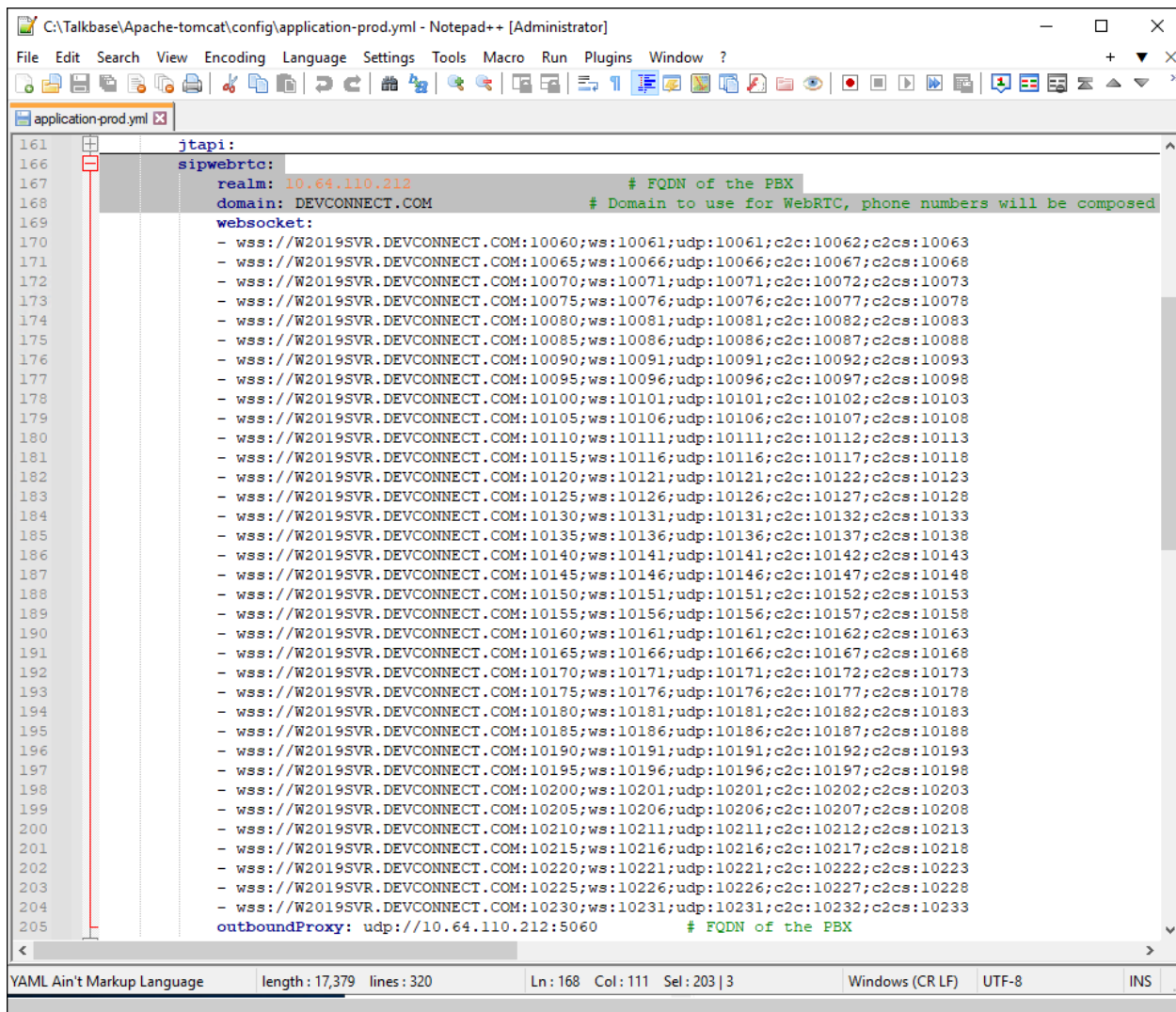
Verify the **sipphonestatus** entries have the Session Manager IP Address (e.g., **10.64.110.212**) for **host**, the SIP Domain from **Section 6.1.1** (e.g., **avaya.com**) for **domain**, and the port number from **Section 6.3** (e.g., **5060**) for **serverport**.



```
81  elasticsearch:
86  Talkbase:
90  talkbase:
91    ui:
95    sites:
99    telephony:
103   search:
105   security:
106     domain: DEVCONNECT.COM           # Active Directory Domain
107     ldap:
115     lyncserver:
126     xmppserver:
144     sipphonestatus:                  #Used for Cisco and Avaya
145       host: 10.64.110.212             # FQDN of the Cisco Unified Communication Manager or t
146       domain: avaya.com                # The SIP domain used for phones
147       clientport: 5065                 # The talkbase side SIP port on which talkbase will listen fo
148       serverport: 5060                 # The PEX side SIP port to use for phone status subscription
```

YAML Ain't Markup Language length: 17,379 lines: 320 Ln: 148 Col: 110 Sel: 550 | 5 Windows (CR LF) UTF-8 INS

Verify the **sipwebrtc** entries have the Session Manager IP Address (e.g., **10.64.110.212**) for **realm** and **outboundProxy**, and the Windows Active Directory domain (e.g., **DEVCONNECT.COM**) for **domain**.



```
161 jtapi:
166 sipwebrtc:
167   realm: 10.64.110.212           # FQDN of the PBX
168   domain: DEVCONNECT.COM       # Domain to use for WebRTC, phone numbers will be composed
169   websocket:
170     - wss://W2019SVR.DEVCONNECT.COM:10060;ws:10061;udp:10061;c2c:10062;c2cs:10063
171     - wss://W2019SVR.DEVCONNECT.COM:10065;ws:10066;udp:10066;c2c:10067;c2cs:10068
172     - wss://W2019SVR.DEVCONNECT.COM:10070;ws:10071;udp:10071;c2c:10072;c2cs:10073
173     - wss://W2019SVR.DEVCONNECT.COM:10075;ws:10076;udp:10076;c2c:10077;c2cs:10078
174     - wss://W2019SVR.DEVCONNECT.COM:10080;ws:10081;udp:10081;c2c:10082;c2cs:10083
175     - wss://W2019SVR.DEVCONNECT.COM:10085;ws:10086;udp:10086;c2c:10087;c2cs:10088
176     - wss://W2019SVR.DEVCONNECT.COM:10090;ws:10091;udp:10091;c2c:10092;c2cs:10093
177     - wss://W2019SVR.DEVCONNECT.COM:10095;ws:10096;udp:10096;c2c:10097;c2cs:10098
178     - wss://W2019SVR.DEVCONNECT.COM:10100;ws:10101;udp:10101;c2c:10102;c2cs:10103
179     - wss://W2019SVR.DEVCONNECT.COM:10105;ws:10106;udp:10106;c2c:10107;c2cs:10108
180     - wss://W2019SVR.DEVCONNECT.COM:10110;ws:10111;udp:10111;c2c:10112;c2cs:10113
181     - wss://W2019SVR.DEVCONNECT.COM:10115;ws:10116;udp:10116;c2c:10117;c2cs:10118
182     - wss://W2019SVR.DEVCONNECT.COM:10120;ws:10121;udp:10121;c2c:10122;c2cs:10123
183     - wss://W2019SVR.DEVCONNECT.COM:10125;ws:10126;udp:10126;c2c:10127;c2cs:10128
184     - wss://W2019SVR.DEVCONNECT.COM:10130;ws:10131;udp:10131;c2c:10132;c2cs:10133
185     - wss://W2019SVR.DEVCONNECT.COM:10135;ws:10136;udp:10136;c2c:10137;c2cs:10138
186     - wss://W2019SVR.DEVCONNECT.COM:10140;ws:10141;udp:10141;c2c:10142;c2cs:10143
187     - wss://W2019SVR.DEVCONNECT.COM:10145;ws:10146;udp:10146;c2c:10147;c2cs:10148
188     - wss://W2019SVR.DEVCONNECT.COM:10150;ws:10151;udp:10151;c2c:10152;c2cs:10153
189     - wss://W2019SVR.DEVCONNECT.COM:10155;ws:10156;udp:10156;c2c:10157;c2cs:10158
190     - wss://W2019SVR.DEVCONNECT.COM:10160;ws:10161;udp:10161;c2c:10162;c2cs:10163
191     - wss://W2019SVR.DEVCONNECT.COM:10165;ws:10166;udp:10166;c2c:10167;c2cs:10168
192     - wss://W2019SVR.DEVCONNECT.COM:10170;ws:10171;udp:10171;c2c:10172;c2cs:10173
193     - wss://W2019SVR.DEVCONNECT.COM:10175;ws:10176;udp:10176;c2c:10177;c2cs:10178
194     - wss://W2019SVR.DEVCONNECT.COM:10180;ws:10181;udp:10181;c2c:10182;c2cs:10183
195     - wss://W2019SVR.DEVCONNECT.COM:10185;ws:10186;udp:10186;c2c:10187;c2cs:10188
196     - wss://W2019SVR.DEVCONNECT.COM:10190;ws:10191;udp:10191;c2c:10192;c2cs:10193
197     - wss://W2019SVR.DEVCONNECT.COM:10195;ws:10196;udp:10196;c2c:10197;c2cs:10198
198     - wss://W2019SVR.DEVCONNECT.COM:10200;ws:10201;udp:10201;c2c:10202;c2cs:10203
199     - wss://W2019SVR.DEVCONNECT.COM:10205;ws:10206;udp:10206;c2c:10207;c2cs:10208
200     - wss://W2019SVR.DEVCONNECT.COM:10210;ws:10211;udp:10211;c2c:10212;c2cs:10213
201     - wss://W2019SVR.DEVCONNECT.COM:10215;ws:10216;udp:10216;c2c:10217;c2cs:10218
202     - wss://W2019SVR.DEVCONNECT.COM:10220;ws:10221;udp:10221;c2c:10222;c2cs:10223
203     - wss://W2019SVR.DEVCONNECT.COM:10225;ws:10226;udp:10226;c2c:10227;c2cs:10228
204     - wss://W2019SVR.DEVCONNECT.COM:10230;ws:10231;udp:10231;c2c:10232;c2cs:10233
205   outboundProxy: udp://10.64.110.212:5060           # FQDN of the PBX
```

YAML Ain't Markup Language length: 17,379 lines: 320 Ln: 168 Col: 111 Sel: 203 | 3 Windows (CR LF) UTF-8 INS

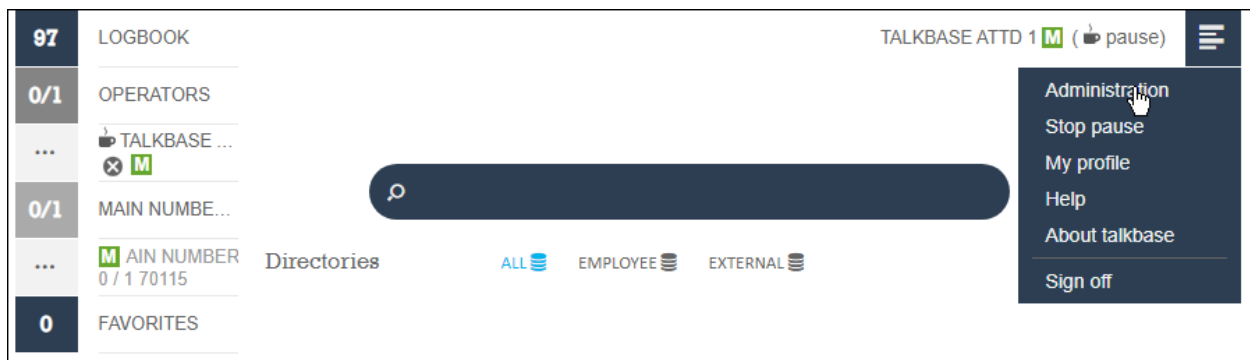
Websocket entries specify WebRTC ports configured for talkbase. Note the ports specified (e.g., ports**10060,10065,10070,10075,10080,10085,10090,10095,10100,10105,10110,10115,10120,10125,10130,10135,10140,10145,10150,10155,10160,10165,10170,10175,10180,10185,10190,10195,10200,10205,10210,10215,10220,10225,10230**). These ports are input to the **Site Detail** tab in **Section 7.3**.

7.3. Configure talkbase Attendant

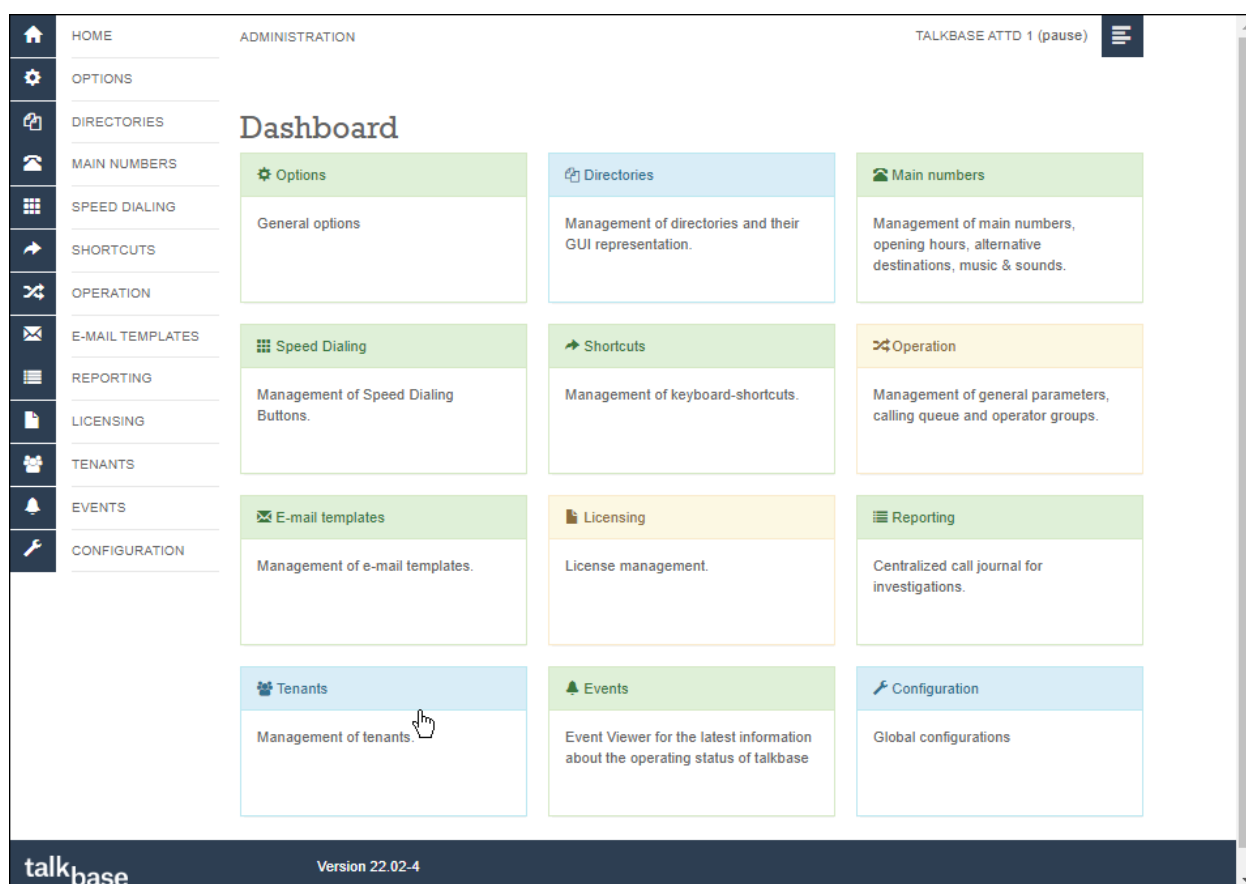
talkbase Attendant administration configures attendants, main numbers, and assigns attendants to answer calls from main numbers. Open a web browser (compliance testing employed Chrome browser) and navigate to talkbase Attendant using the URL **https://<FQDN of talkbase server>**. Enter the Active Directory user credentials of a user with **Talkbase_Admin** and **Talkbase_SuperUser** group memberships and click on **Sign in**.

The image shows the 'talkbase Sign Up' page. At the top right is the 'talkbase' logo. Below it, the text 'Sign Up' is prominently displayed. Underneath, there is a 'Sign-in address' label followed by a text input field containing the placeholder 'Your sign-in address'. Below that is a 'Password' label followed by a text input field containing the placeholder 'Your password'. A checkbox labeled 'Automatic Login' is checked. At the bottom left of the form is a 'Sign In' button. The 'talkbase' logo is also present in the bottom left corner of the page's footer area.

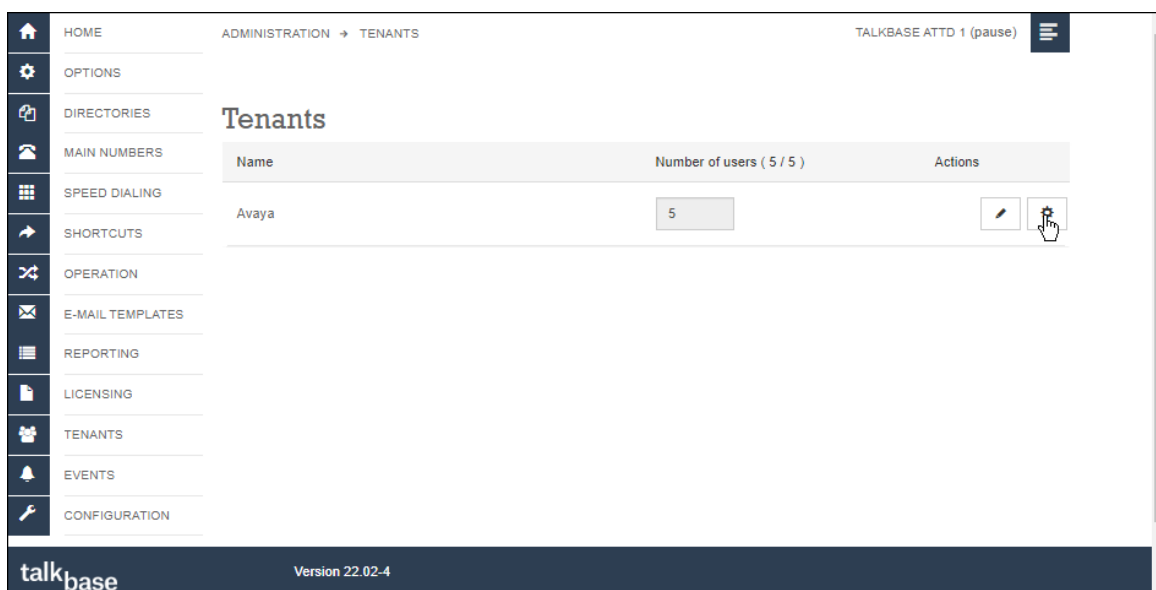
Click on the icon at the top right of the page and select **Administration** as shown below.



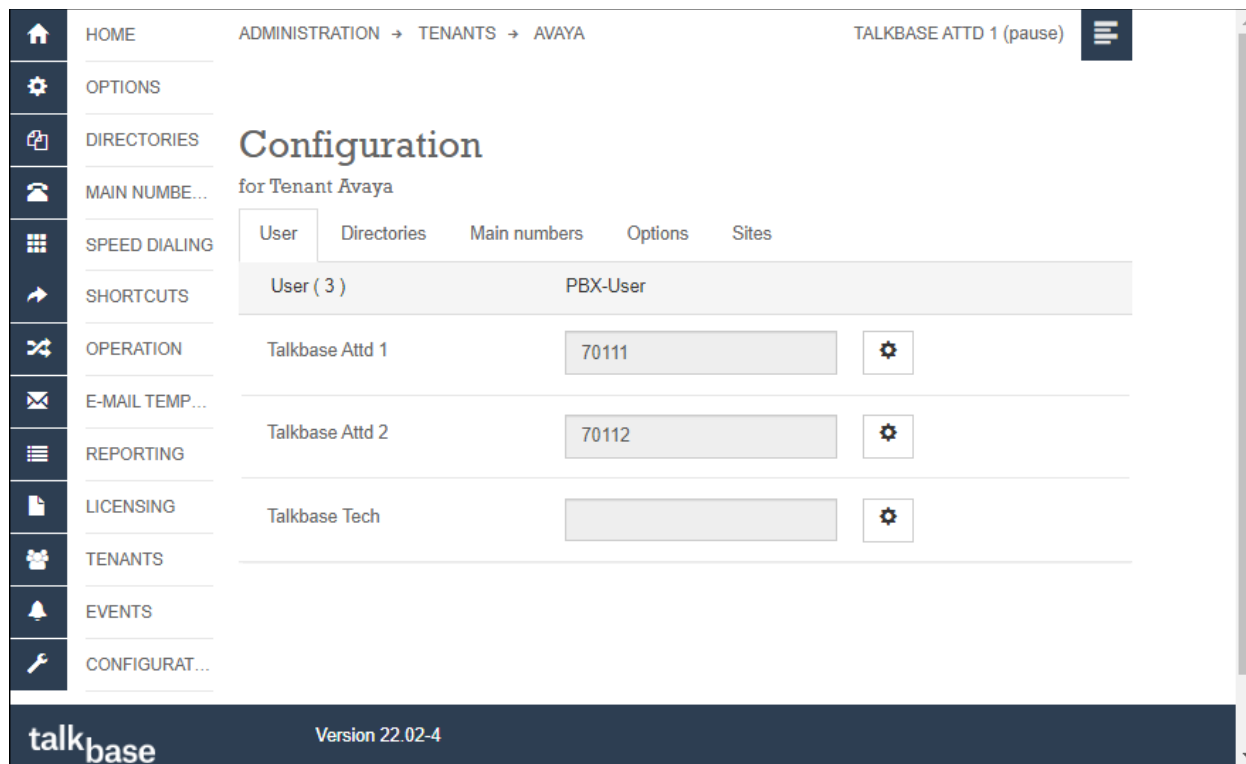
The screen displays the various modules. Click on **Tenants** at the bottom of the screen.



The **Tenant** called **Avaya** is set up by the FROX engineers as part of the original connection setup. Click on the detail (gear) icon to display the information on this tenant.



The users are automatically populated from the Active Directory of the domain that the talkbase server is a part of. **Talkbase Attdd 1**, **Talkbase Attdd2** and **Talkbase Tech** are all added as domain users as per **Section 7.1**. Talkbase Attendant users are shown with **PBX-Users** assigned. To assign **PBX-User**, click on the **Interface Configuration** (gear) icon.



HOME ADMINISTRATION → TENANTS → AVAYA TALKBASE ATTDD 1 (pause)

OPTIONS

DIRECTORIES

MAIN NUMBE...

SPEED DIALING

SHORTCUTS

OPERATION

E-MAIL TEMP...

REPORTING

LICENSING

TENANTS

EVENTS

CONFIGURAT...

Configuration

for Tenant Avaya

User Directories Main numbers Options Sites

User (3) PBX-User

Talkbase Attdd 1	70111	
Talkbase Attdd 2	70112	
Talkbase Tech		

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Enter the SIP login name and Communication Profile Password from **Section 6.6** for **PBX-User** and **PBX-Password**. The **Phone number** is the CM Endpoint Profile Extension of the SIP user.

PBX-User

PBX-User

Phone number

PBX-Password

Confirm Password

70111

70111

.....

.....

Save

Cancel

After configuring Users, select the **Sites** tab. The site should be configured so a Main Number

can be assigned. To create a site, click the + icon. Compliance testing used a **Name** of **Avaya**, **SIP Domain** of **avaya.com**, and set **Default** (not shown).

The screenshot displays the Talkbase Administration web interface. The left sidebar contains a navigation menu with icons and labels for various system functions: HOME, OPTIONS, DIRECTORIES, MAIN NUMBE..., SPEED DIALING, SHORTCUTS, OPERATION, E-MAIL TEMP..., REPORTING, LICENSING, TENANTS, EVENTS, and CONFIGURAT... The main content area shows the breadcrumb path 'ADMINISTRATION → TENANTS → AVAYA' and the status 'TALKBASE ATTD 1 (pause)'. The title 'Configuration for Tenant Avaya' is prominently displayed. Below the title, there are tabs for 'User', 'Directories', 'Main numbers', 'Options', and 'Sites', with 'Sites' being the active tab. A table lists the configured sites with columns for 'Default', 'Name', 'SIP Domain', and 'Actions'. One site is listed: 'Avaya' with 'avaya.com' as the SIP Domain. An edit icon (pencil) is visible in the Actions column. A '+ ' button is located at the bottom right of the table area to add new sites. The footer shows the 'talkbase' logo and 'Version 22.02-4'.

Default	Name	SIP Domain	Actions
<input checked="" type="checkbox"/>	Avaya	avaya.com	

Select the **Edit** icon for the Avaya Site. Once open, select the **Detail** tab. Click the **Edit** icon to configure the following. The **PBX-IP or FQDN** should be set to the Session Manager IP Address (e.g., **10.64.110.212**) Set the **WebRTC FQDN** to the talkbase server FQDN (e.g., **W2019SVR.DEVCONNECT.COM**). Set the **Realm** to the SIP domain from **Section 6.1** (e.g., **avaya.com**). Add the **WebRTC ports** noted from **Section 7.2**. Set the **Protocol for WebRTC** to **UDP**. Set the **SIP Port** to **5060**.

The screenshot displays the 'Site "Avaya"' configuration page in the talkbase administration interface. The interface includes a left sidebar with navigation icons and labels, a top navigation bar, and a main content area. The 'Detail' tab is selected for the 'Avaya' site.

Navigation Bar: HOME | ADMINISTRATION → TENANTS → AVAYA → AVAYA | TALKBASE ATTD 1 (pause) | Menu icon

Left Sidebar: HOME, OPTIONS, DIRECTORIES, MAIN NUMBE..., SPEED DIALING, SHORTCUTS, OPERATION, E-MAIL TEMP..., REPORTING, LICENSING, TENANTS, EVENTS, CONFIGURAT...

Site "Avaya" Configuration:

- PBX-IP or FQDN:** 10.64.110.212
- Priority:** 1
- WebRTC FQDN:** W2019SVR.DEVCONNECT.COM
- Realm:** avaya.com
- WebRTC ports:** 10060,10065,10070,10075,10080,10085,10090,10095,10100,1
- Protocol for WebRTC:** UDP
- SIP Port:** 5060

Footer: talkbase | Version 22.02-4

Return to the **Administration** menu and select **Main numbers** from the **Dashboard** (not shown) to configure the main numbers that will be assigned to attendants. The following shows that two numbers (e.g., **70115** and **70116**) are configured for the Avaya Tenant. To add a main number, Click on the + icon to add a new row and input an extension that has been configured to route from Session Manager to talkbase and select the **Avaya** Tenant.

The screenshot displays the 'Main numbers' configuration page in the talkbase Administration interface. The left sidebar contains navigation links: HOME, OPTIONS, DIRECTORIES, MAIN NUMBERS (selected), SPEED DIALING, SHORTCUTS, OPERATION, E-MAIL TEMPLATES, REPORTING, LICENSING, TENANTS, EVENTS, and CONFIGURATION. The top navigation bar shows 'ADMINISTRATION → MAIN NUMBERS' and 'TALKBASE ATTD 1 (pause)'. The main content area is titled 'Main numbers' and includes a dropdown menu for 'Use this number for outgoing calls:' set to 'Main number / 70115'. Below this is a table with columns: Active, Name, Number, Priority, Color, Icon, and Site. Two rows are listed: 'Main number' with number 70115 and 'Main Number 2' with number 70116, both assigned to the 'Avaya' site. A '+' icon is present to add new numbers. Below the table, a section titled 'Internal/external classification of phone numbers' contains a note about configuration rules for 'I' (one or more numbers), 'x' (exactly one number), and '+' (in front of numbers).

Active	Name	Number	Priority	Color	Icon	Site
<input checked="" type="checkbox"/>	Main number	70115	99	#23	<input checked="" type="checkbox"/>	Avaya
<input checked="" type="checkbox"/>	Main Number 2	70116	99	#23	<input checked="" type="checkbox"/>	Avaya

Internal/external classification of phone numbers

The internal/external assignment of phone numbers can be configured using the following rules: 'I' = one or more numbers, 'x' = exactly one number, 'I' (e.g. [1234]) a selection of numbers, a '+' in front of numbers

Scroll down to add **Internal/external classification of phone numbers** to classify incoming calls to talkbase as internal or external. Click on the + icon to add a new row. Input the **Rule** and specify the classification as **Internal** or **External**. Priority can be adjusted by moving the rule up or down by clicking the arrow icons for the row. The following rules are added for compliance testing. 5-digit extensions are classified as internal.

The screenshot displays the 'Internal/external classification of phone numbers' configuration page in the talkbase system. The page features a sidebar on the left with navigation links: OPERATION, E-MAIL TEMP..., REPORTING, LICENSING, TENANTS, EVENTS, and CONFIGURAT... The main content area is titled 'Internal/external classification of phone numbers' and includes an informational message: 'The internal/external assignment of phone numbers can be configured using the following rules: '!' = one or more numbers, 'x' = exactly one number, '['' (e.g. [1234]) a selection of numbers, a '+' in front of numbers'. Below this, a table lists the configured rules:

Priority	Rule	Classification as an internal or external number?	Actions
1	xxxxx	<input type="radio"/> External <input checked="" type="radio"/> Internal	Up Arrow, Down Arrow, Delete
2	!	<input checked="" type="radio"/> External <input type="radio"/> Internal	Up Arrow, Down Arrow, Delete

A plus icon (+) is located at the bottom right of the table to add new rules. The footer of the interface shows the 'talkbase' logo and 'Version 22.02-4'.

Return to the **Administration** menu and select **Operation** from the **Dashboard** (not shown) to assign attendants to receive calls from main numbers. The **Operator Groups** tab and **Main numbers** subtabs show numbers configured along with attendants administered. **Overflow** shows the number of calls in queue needed before another **Operator group** will receive calls from the main number. Compliance testing administered **Talkbase Attd 1** and **Talkbase Attd 2** to different main numbers as seen below.

ADMINISTRATION → OPERATION TALKBASE ATTD 1 (pause)

Operation

Operator groups Queue configuration

	70115	70116
Main number	70115	
Overflow	2 2 2 2 2 2	
Operator group	- 1 2 3 4 5 6	
Talkbase Attd 1	<input type="radio"/> <input checked="" type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/>	
Talkbase Attd 2	<input checked="" type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/>	
Talkbase Tech	<input checked="" type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/>	

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ADMINISTRATION → OPERATION TALKBASE ATTD 1 (pause)

Operation

Operator groups Queue configuration

	70115	70116
Main number		70116
Overflow		5 5 5 5 5 5
Operator group		- 1 2 3 4 5 6
Talkbase Attd 1		<input checked="" type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/>
Talkbase Attd 2		<input type="radio"/> <input checked="" type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/>
Talkbase Tech		<input checked="" type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/> <input type="radio"/>

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The example below shows both **Talkbase Att'd 1** and **Talkbase Att'd 2** assigned to the **Main number 70115**.

HOME

ADMINISTRATION → OPERATION

TALKBASE ATTD 1 (pause)

OPTIONS

DIRECTORIES

MAIN NUMBERS

SPEED DIALING

SHORTCUTS

OPERATION

E-MAIL TEMPLATES

REPORTING

LICENSING

TENANTS

EVENTS

CONFIGURATION

Operation

Operator groups

Queue configuration

70115

70116

Main number

70115

Overflow

2

2

2

2

2

2

Operator group

-

1

2

3

4

5

6

Talkbase Att'd 1

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Talkbase Att'd 2

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Talkbase Tech

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talkbase

Version 22.02-4

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of talkbase with Communication Manager and Session Manager.

8.1. Verify the SIP Trunk connection

The SIP trunk connection can be verified from both Communication Manager and Session Manager.

8.1.1. Verify Avaya Aura® Communication Manager Trunk

From the SAT interface, verify the status of the SIP trunk groups by using the **status trunk n** command, where “n” is the trunk group number administered in **Section 5.3**. Verify that all trunks are in the **in-service/idle** state.

```
status trunk 12
                                TRUNK GROUP STATUS

Member      Port      Service State      Mtce Connected Ports
              Busy

0012/0001 T000026 in-service/idle no
0012/0002 T000027 in-service/idle no
0012/0003 T000028 in-service/idle no
0012/0004 T000029 in-service/idle no
0012/0005 T000030 in-service/idle no
0012/0006 T000031 in-service/idle no
0012/0007 T000032 in-service/idle no
0012/0008 T000033 in-service/idle no
0012/0009 T000034 in-service/idle no
0012/0010 T000035 in-service/idle no
```

Verify the status of the SIP signaling groups by using the **status signaling-group n** command, where “n” is the signaling group number administered in **Section 5.3**. Verify that the signaling group is **in-service** as indicated in the **Group State** field.

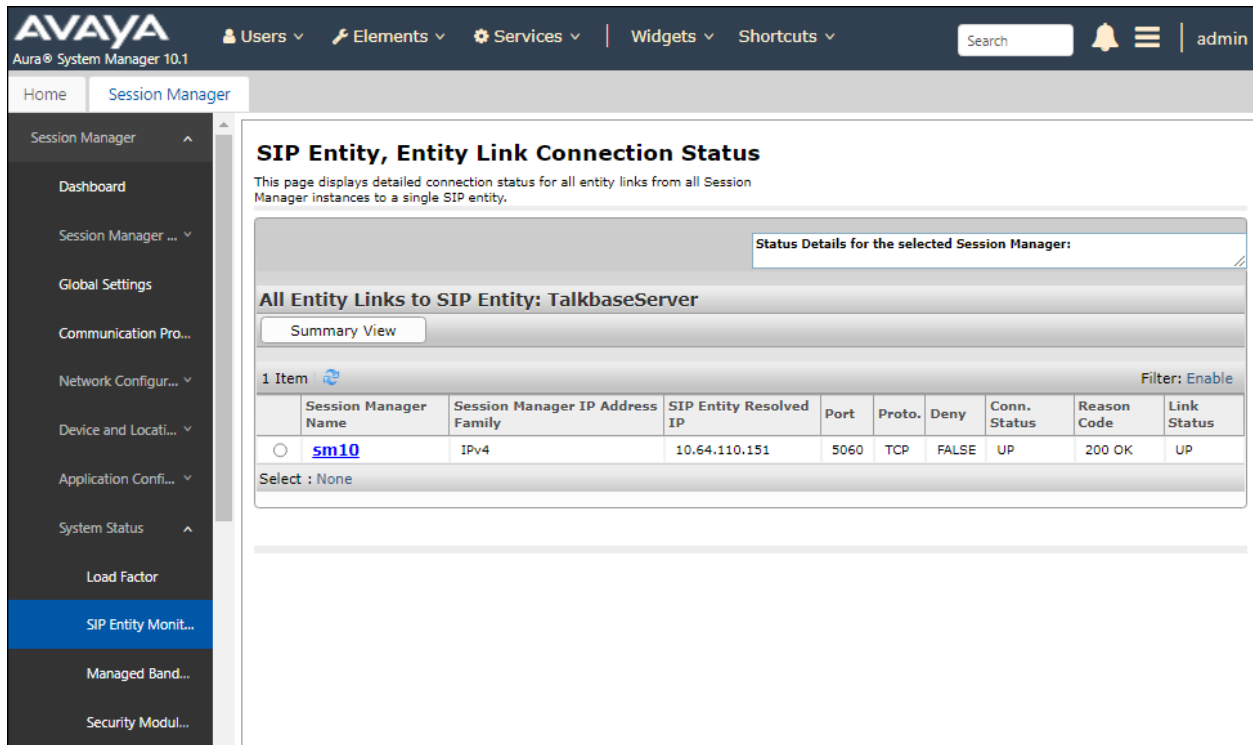
```
status signaling-group 12
                        STATUS SIGNALING GROUP

      Group ID: 12
      Group Type: sip

      Group State: in-service
```

8.2. Verify talkbase SIP Entity is Connected

Log into System Manager as per **Section 6**. Navigate to **Elements → Session Manager→System Status→ SIP Entity Monitoring**. Click on the name of the talkbase SIP entity created in **Section 6.2** e.g., **TalkbaseServer** (not shown). The SIP Entity Link's **Conn. Status** should show **UP**.

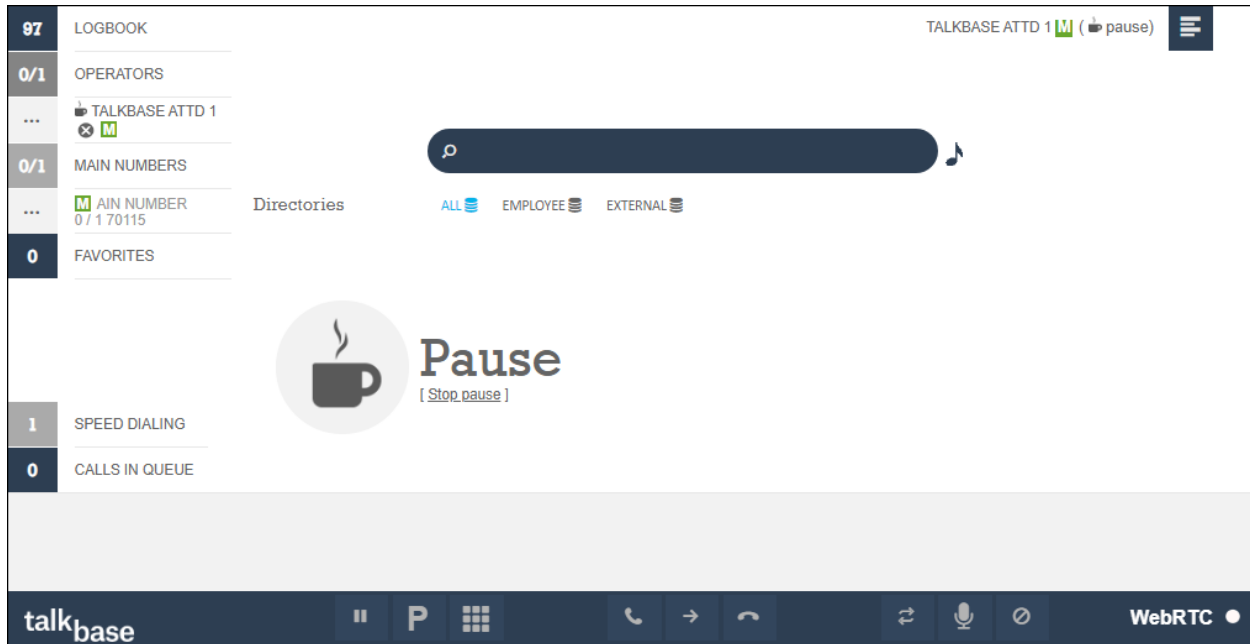


The screenshot shows the Avaya Aura System Manager 10.1 interface. The top navigation bar includes the Avaya logo, a search bar, and links for Users, Elements, Services, Widgets, and Shortcuts. The left sidebar contains a menu with options like Session Manager, Dashboard, Global Settings, and SIP Entity Monitoring. The main content area is titled "SIP Entity, Entity Link Connection Status" and displays a table of entity links for the selected Session Manager (sm10). The table shows one item with a status of UP.

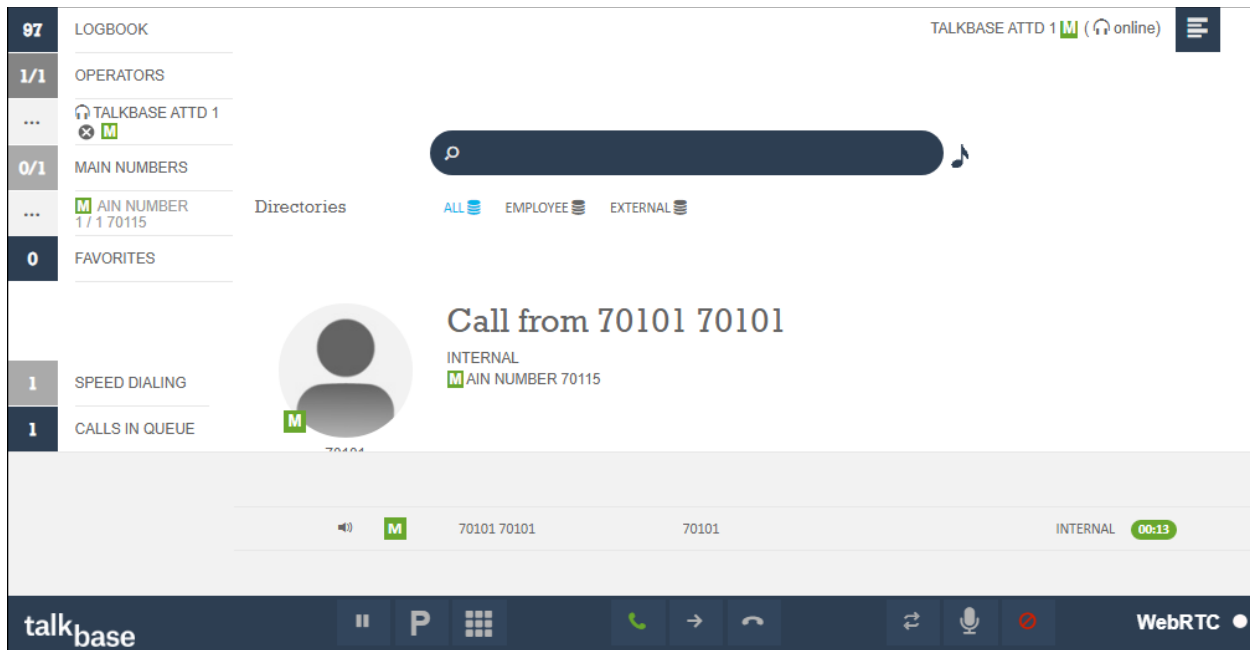
Session Manager Name	Session Manager IP Address	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
sm10	IPv4	10.64.110.151	5060	TCP	FALSE	UP	200 OK	UP

8.3. Verify talkbase Attendant

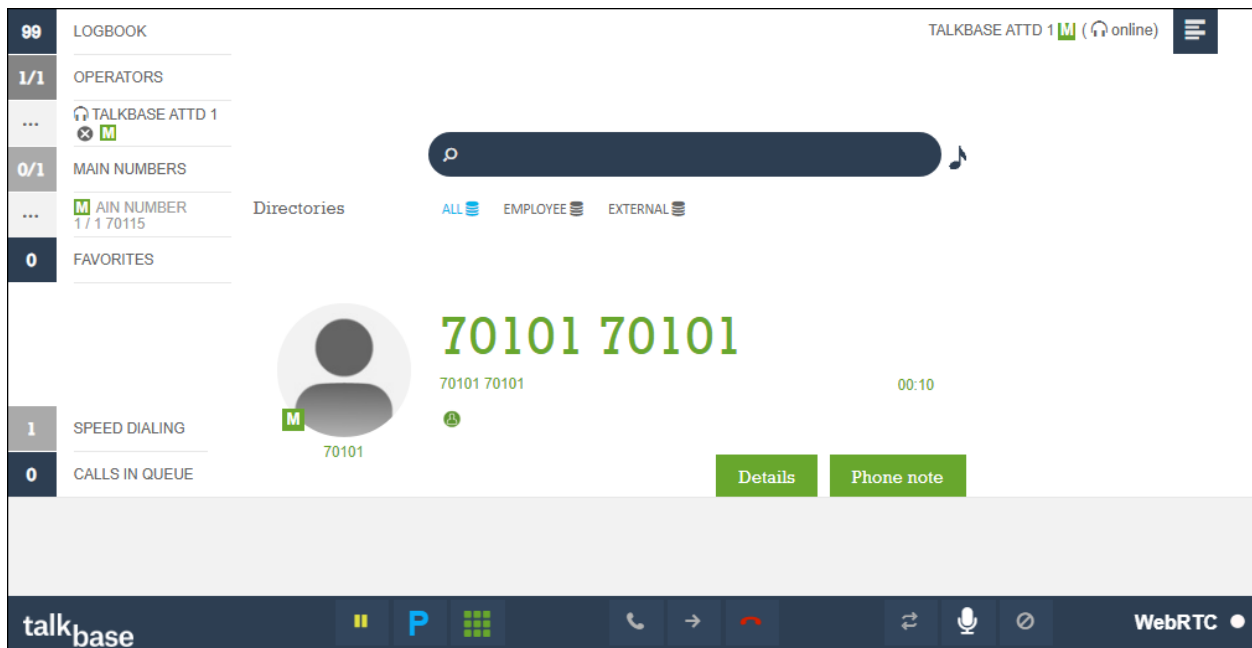
Open a web connection to the talkbase server as per **Section 7.2**. Enter the appropriate credentials and click on **Sign in** (not shown). The following screen should display showing the agent and the message **Pause** showing that the Attendant is not idle. **WebRTC** displayed in white at the bottom of the screen indicates that the Attendant is connected correctly with the SIP user created in **Section 6.6**.



Clicking on **Pause** will place the Attendant into a state ready to receive calls. When a call from **70101** is placed to the main number **70115** the following should display as shown below. Clicking on either the message in the middle of the screen or on the green telephone icon at the very bottom of the screen will answer the call.



Once the call is answered, verify the call can be put on hold, forwarded, or ended using the icons located at the bottom of the screen. The CLID displayed in the middle of the screen turns green also indicating the call is answered.



After answering the call, verify two-way audio.

9. Conclusion

These Application Notes describe the configuration steps required to integrate FROX AG talkbase with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

10. Additional References

This section references the product documentation that is relevant to these Application Notes. The following Avaya product documentation is available at support.avaya.com.

- [1] *Administering Avaya Aura® Communication Manager*, Release 10.1.x, Issue 2, September 2022.
- [2] *Administering Avaya Aura® Session Manager*, Release 10.1, Issue 4, September 2022.

The following documentation related to talkbase may be obtained from Frox AG.

- [3] Talkbase Administration Guide, Release 2.17, January 4, 2019
- [4] Talkbase Installation Guide, Release 22.02, September 22, 2022
- [5] Talkbase Operator's Guide, Release 1.72, February 8, 2017
- [6] Talkbase Prerequisites, Release 22.02, August 25, 2022

11. Appendix 1 Talkbase server Configuration File

The talkbase server configuration file can be modified for specific configurations as illustrated in **Section 7.2**. The configuration file used for compliance testing is listed here excepting password credentials.

```
server:
  port: 8080
  address: localhost
  servlet:
    session:
      timeout: 180 # Session timeout. If a duration suffix is not specified, seconds will
be used.

spring:
  profiles: prod
  datasource:
    type: com.zaxxer.hikari.HikariDataSource
    url: jdbc:postgresql://localhost:5432/talkbase
    name:
    username: talkbase
    password: *****
    driver-class-name: org.postgresql.Driver
    statementCacheNumDeferredCloseThreads: 1
    unreturnedConnectionTimeout: 180
    debugUnreturnedConnectionStackTraces: true
    maximumPoolSize: 150
  jpa:
    database-platform: com.noser.hunter.common.util.FixedPostgreSQL82Dialect
    database: POSTGRESQL
    show-sql: false
    properties:
      hibernate.cache.use_second_level_cache: true
      hibernate.cache.use_query_cache: false
      hibernate.generate_statistics: true
      hibernate.javax.cache.provider: org.ehcache.jsr107.EhcacheCachingProvider
      hibernate.javax.cache.uri: ${spring.cache.jcachel.config}
      hibernate.cache.region.factory_class:
com.noser.hunter.common.util.SpringCacheRegionFactory
#
#   datasource:
#     dataSourceClassName: com.microsoft.sqlserver.jdbc.SQLServerDataSource
#     url: jdbc:sqlserver://localhost:1434;databaseName=talkbase;trustServerCertificate=true
#     databaseName:
#     serverName:
#     username: sa
#     password: *****
#     maximumPoolSize: 100
#     driver-class-name: com.microsoft.sqlserver.jdbc.SQLServerDriver
#   jpa:
#     database-platform: org.hibernate.dialect.SQLServer2012Dialect
#     database: SQL_SERVER
#     openInView: false
#     show_sql: false
#     generate-ddl: false
#     hibernate:
#       ddl-auto: none
#       naming-strategy: org.hibernate.cfg.EJB3NamingStrategy
#     properties:
#       hibernate.cache.use_second_level_cache: false
#       hibernate.cache.use_query_cache: false
#       hibernate.generate_statistics: false
#       hibernate.cache.region.factory_class:
org.hibernate.cache.ehcache.SingletonEhCacheRegionFactory
cache:
  jcachel:
    config: ehcache.xml

thymeleaf:
```



```

mode: XHTML
cache: true

mail:
  host: mail.tblab.frox.com
  port: 25
  user: TBTech
  password: *****
  from: TBTech@DEVCONNECT.COM
  protocol: smtp
  tls: false
  auth: false

metrics:
  jmx.enabled: true
  graphite:
    enabled: false
    host: localhost
    port: 2003
    prefix: test

elasticsearch:
  host: 127.0.0.1
  port: 9300
  adminPort: 9200
  clusterName: elasticsearch
Talkbase:
  presence: none
  callForwarding: avaya

talkbase:
  ui:
    company:
      toolbar_logo: # Absolute path to custom company logo (i.e. D:/toolbar-logo.jpg,
                    # JPG/JPEG formats are supported, preferred size - 30 x 30) that display on the main screen in the
                    # bar at the bottom of the screen left to the Talkbase logo
      login_logo: # Absolute path to custom logo (i.e. D:/login-logo.jpg, JPG/JPEG
                  # formats are supported, preferred size - 75 x 75) that should display on the login screen near
                  # Talkbase logo
    sites:
      configuration:
        path: ../sitesConfiguration # Path to Talkbase tenant sites configuration
file
      file: sites-config.json # The name of the Talkbase tenant sites
configuration file
      telephony:
        vendor: avaya # Possible values: skype_for_business, cisco, avaya,
cloud_pbx, innovaphone
        pbx: 10.64.110.212 # Comma-separated list of FQDNs or IP
        # addresses of one or more PBXes to which talkbase can connect (empty for Skype for Business)
        sipdomain: avaya.com # SIP domain to be used for outgoing calls
      search:
        path: ../elasticsearch # Path to the Elastic Search Engine
      security:
        domain: DEVCONNECT.COM # Active Directory Domain
      ldap:
        globalCatalogPort: 3268 # default(3268) for SSL (3269) Active Directory
Search Port (Global Catalog)
        ldapPort: 389 # default(389) for SSL (636)
        prefix: ldap # ldaps for SSL
        adFQDN: # If this property is set it will be taken as the
FQDN of your Active Directory server
        account:
          user: TBTech # Technical Active Directory Username
          password: ***** # Password of the technical AD user
      lyncserver:
        domain: DEVCONNECT.COM # Domain of the Skype for Business Server
        sipdomain: DEVCONNECT.COM # Used for autodiscovering Skype for Business
(lyncdiscoverinternal must be defined in DNS)
        techauth_use_domain: true

```

```

mediation_server:          # Pool FQDN of the mediation server; leave empty
to let talkbase autodiscover it
account:
    user: TBTech           # Technical user to use for Skype for Business
Authentication
    password: *****    # Password of the technical user
    localurl: localhost    # Should be the local host IP address
    port: 5074             # Configured Skype for Business SIP server port
allowInsecureConnection: true # This property is needed if there's no valid
certificate for https
xmpps_server:
    enabled: false
    host: 0.0.0.0          # FQDN of the presence server
    port: 5222             # XMPP port (default: 5222)
    serviceName: DEVCONNECT.COM # presence domain as configured on the presence
server (everything after @)
    imUserFormat: pbxUser    #pbxUser or phoneNumber
    autodiscoverDomain: DEVCONNECT.COM # used for autodiscover if empty the security
domain is fallback
    imKeyField: email_address # Which DB field contains the IM & Presence
Usernames? Default: ad_username Possible values: ad_username, user_principal_name, email_address,
sip_address, jabber_id, im_username
    adPresenceKey: email     # Which AD field contains the IM & Presence
Usernames of Operator? Default: sAMAccountName
    useAvayaPresenceSymbols: false
    account:
        user: username      # id of the XMPP user (Cisco End User or Avaya
presence user )
        password: *****  # password of XMPP user
        resource: Talkbase  # Informational name of the talkbase resource
        reconnectionDelay: 120 # fixed delay in seconds between the XMPP reconnection
attempts
        connectionRetryDelay: 60 # fixed delay in seconds for establishing XMPP connection
if initial was failed
        replyTimeout: 3000    # number of milliseconds to wait for a response from the
XMPP server
        fixedRateToActualizeRoster: 300000 # fixed period in milliseconds between invocations
siphonestatus:
    host: 10.64.110.212    # FQDN of the Cisco Unified Communication
Manager or the Avaya Session Manager SIP Entity
    domain: avaya.com      # The SIP domain used for phones
    clientport: 5065       # The talkbase side SIP port on which talkbase
will listen for status changes, default 5065
    serverport: 5060       # The PBX side SIP port to use for phone status
subscriptions, default 5060
    account:
        username: 70110    # User name of technical user configured on the PBX
        password: ***** # Password of the technical user
        phonefields: phone # Which db fields should be considered for phoneStatus (order
matters) example : phone, phone2, phone3, mobile
    sippresence:
        host: 10.64.110.212 # FQDN or IP address of the PBX
        clientport: 5065    # Port to be used on the talkbase server (default
5065)
        serverport: 5060    # Port to be used on the PBX (default 5060)
        imKeyField: ad_username # Which DB field contains the IM & Presence
Usernames? Default: ad_username Possible values: ad_username, user_principal_name, email_address,
sip_address, jabber_id, im_username, phone
    account:
        username: TBTech    # User name of technical user configured on the PBX
        password: *****  # Password of the technical user
    jtapi:
        host: 10.64.110.212 # FQDN of the Cisco Unified Communication
Manager
        account:
            user:           # User name of the technical application user (this is an Application
User on the CUCM)
            password:       # Password of the technical application user
        sipwebrtc:
            realm: 10.64.110.212 # FQDN of the PBX

```

```

domain: DEVCONNECT.COM # Domain to use for WebRTC, phone numbers will
be composed like <phoneNumber>@<domain>
websocket:
- wss://W2019SVR.DEVCONNECT.COM:10060;ws:10061;udp:10061;c2c:10062;c2cs:10063
- wss://W2019SVR.DEVCONNECT.COM:10065;ws:10066;udp:10066;c2c:10067;c2cs:10068
- wss://W2019SVR.DEVCONNECT.COM:10070;ws:10071;udp:10071;c2c:10072;c2cs:10073
- wss://W2019SVR.DEVCONNECT.COM:10075;ws:10076;udp:10076;c2c:10077;c2cs:10078
- wss://W2019SVR.DEVCONNECT.COM:10080;ws:10081;udp:10081;c2c:10082;c2cs:10083
- wss://W2019SVR.DEVCONNECT.COM:10085;ws:10086;udp:10086;c2c:10087;c2cs:10088
- wss://W2019SVR.DEVCONNECT.COM:10090;ws:10091;udp:10091;c2c:10092;c2cs:10093
- wss://W2019SVR.DEVCONNECT.COM:10095;ws:10096;udp:10096;c2c:10097;c2cs:10098
- wss://W2019SVR.DEVCONNECT.COM:10100;ws:10101;udp:10101;c2c:10102;c2cs:10103
- wss://W2019SVR.DEVCONNECT.COM:10105;ws:10106;udp:10106;c2c:10107;c2cs:10108
- wss://W2019SVR.DEVCONNECT.COM:10110;ws:10111;udp:10111;c2c:10112;c2cs:10113
- wss://W2019SVR.DEVCONNECT.COM:10115;ws:10116;udp:10116;c2c:10117;c2cs:10118
- wss://W2019SVR.DEVCONNECT.COM:10120;ws:10121;udp:10121;c2c:10122;c2cs:10123
- wss://W2019SVR.DEVCONNECT.COM:10125;ws:10126;udp:10126;c2c:10127;c2cs:10128
- wss://W2019SVR.DEVCONNECT.COM:10130;ws:10131;udp:10131;c2c:10132;c2cs:10133
- wss://W2019SVR.DEVCONNECT.COM:10135;ws:10136;udp:10136;c2c:10137;c2cs:10138
- wss://W2019SVR.DEVCONNECT.COM:10140;ws:10141;udp:10141;c2c:10142;c2cs:10143
- wss://W2019SVR.DEVCONNECT.COM:10145;ws:10146;udp:10146;c2c:10147;c2cs:10148
- wss://W2019SVR.DEVCONNECT.COM:10150;ws:10151;udp:10151;c2c:10152;c2cs:10153
- wss://W2019SVR.DEVCONNECT.COM:10155;ws:10156;udp:10156;c2c:10157;c2cs:10158
- wss://W2019SVR.DEVCONNECT.COM:10160;ws:10161;udp:10161;c2c:10162;c2cs:10163
- wss://W2019SVR.DEVCONNECT.COM:10165;ws:10166;udp:10166;c2c:10167;c2cs:10168
- wss://W2019SVR.DEVCONNECT.COM:10170;ws:10171;udp:10171;c2c:10172;c2cs:10173
- wss://W2019SVR.DEVCONNECT.COM:10175;ws:10176;udp:10176;c2c:10177;c2cs:10178
- wss://W2019SVR.DEVCONNECT.COM:10180;ws:10181;udp:10181;c2c:10182;c2cs:10183
- wss://W2019SVR.DEVCONNECT.COM:10185;ws:10186;udp:10186;c2c:10187;c2cs:10188
- wss://W2019SVR.DEVCONNECT.COM:10190;ws:10191;udp:10191;c2c:10192;c2cs:10193
- wss://W2019SVR.DEVCONNECT.COM:10195;ws:10196;udp:10196;c2c:10197;c2cs:10198
- wss://W2019SVR.DEVCONNECT.COM:10200;ws:10201;udp:10201;c2c:10202;c2cs:10203
- wss://W2019SVR.DEVCONNECT.COM:10205;ws:10206;udp:10206;c2c:10207;c2cs:10208
- wss://W2019SVR.DEVCONNECT.COM:10210;ws:10211;udp:10211;c2c:10212;c2cs:10213
- wss://W2019SVR.DEVCONNECT.COM:10215;ws:10216;udp:10216;c2c:10217;c2cs:10218
- wss://W2019SVR.DEVCONNECT.COM:10220;ws:10221;udp:10221;c2c:10222;c2cs:10223
- wss://W2019SVR.DEVCONNECT.COM:10225;ws:10226;udp:10226;c2c:10227;c2cs:10228
- wss://W2019SVR.DEVCONNECT.COM:10230;ws:10231;udp:10231;c2c:10232;c2cs:10233
outboundProxy: udp://10.64.110.212:5060 # FQDN of the PBX

atiras:
  atiras_interface_enabled: false
  host: <atiras_server_fqdn_or_ip_address>
  port: 8089

ivr:
  audioContainer: C:/Talkbase/IVR/Container/prompt/ulaw # do not change, setup
installs the files there
  audioSubPath: talkbase # do not change, setup
installs the files there
  audioQueueContainer: C:/Talkbase/IVR/Container/queue/ulaw # do not change, setup
installs the files there
  servicenumber:
  soundContainer: ../sounds # do not change, setup
installs the files there
  licencing:
  path: ../licence # do not change, setup
installs the files there
  file: talkbase-licence.dat # the name of the
talkbase license file
  exchange:
  enabled: true # true or false, if true,
exchange is connected at startup
  url: #https://exchangeserver.contoso.com/EWS/Exchange.asmx
proxy:
  url:
  domain:
  port:
  user:
  password:
account:
  domain: DEVCONNECT.COM
  user: TBTech

```

```

        password: *****
        email: TBTech@DEVCONNECT.COM
        timewindow: 30 # in days (max 41)
        amountOfEvents: 15
        timeout: 100000 # milliseconds
exchange_online:
    enabled: false
    client_id:
    client_secret:
    tenant_id:
    eMail_domains:
        - frox.ch
        - frox.com
ldap:
    configuration_folder_path: C:\Talkbase\LdapConfiguration
    page_size: 1000
    schedule: 0 0 1 ? * *
    use_change_notifications: false
csvImport:
    location: C:\Talkbase\CsvFiles
    charSet:
    numberOfBCKs: 5
    schedule: 0 10 2 * * *
    delimiterChar: ';'
    enclosingChar: '"'
    dateFormat: 'dd.MM.yyyy HH:mm:ss'
    supportedFileTypes: csv
logCleanup:
    logDirectories:
        - C:\Talkbase\Logs
        - C:\Talkbase\apache-tomcat\logs
        - C:\Talkbase\elasticsearch\logs
    deleteAfterDays: 30
    taskSchedule: 0 0 1 * * *
voicerecording:
    voiceRecordingDirectory: C:\Talkbase\WebRTC\RecordedCalls
    deleteMarkedCallsAfterDays: 7
    deleteUnmarkedCallsAfterDays: 0
    cleanup:
        schedule: 00 30 1 * * *
innovaphone:
    username: TBTech
    password: *****
    userkey: ad_username # Default: ad_username Possible values: ad_username,
user_principal_name, email_address, sip_address, jabber_id, im_username
    host:
avaya:
    callforwarding:
        aesServer: https://10.64.110.247 # The FQDN of the
Avaya AES Server
        username: TBTech # The CM user (must be able to access
SAT
        password: ***** # The CM user password
        request.timeout: 5000
breather:
    extendDuration: 10 # amount of seconds the breather will be extended by
    maxExtensionsCount: 3 # amount of times the breather can be extended
cache:
    timeToLiveSeconds: 3600
ehcache:
    maxBytesLocalHeap: 512M
    directoryEntryImages:
        maxSizeMemory: 256
        maxSizeDisk: 1024
        timeToLiveInCache: 3600
    audioFiles:
        maxSizeMemory: 128
        maxSizeDisk: 1024
        timeToLiveInCache: 3600

```

```

attendantStatistics:                                     # do not change, it is
installed like this by AttendantStatisticsSetup.exe
  enabled: false
  datasource:
    dataSourceClassName: org.postgresql.ds.PGSimpleDataSource
    url: jdbc:postgresql://localhost:5431/AttendantStatistics
    databaseName:
    serverName:
    username: postgres
    password: *****
    maximumPoolSize: 10
  jpa:
    database-platform: com.noser.hunter.common.util.FixedPostgreSQL82Dialect
    database: POSTGRESQL
    show-sql: false

subscription:
  polling: # not for exchange online
    schedule: 0 * * * * *
  favorites:
    polling:
      schedule: 0 */5 * * * *

```

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