



Application Notes for configuring Teldat VyDa Media Gateway with Avaya Aura[®] Communication Manger R6.0.1 using H.323 Signalling - Issue 1.0

Abstract

These Application Notes describe the configuration steps for provisioning Teldat's VyDa Media Gateway to successfully interoperate with Avaya Aura[®] Communication Manager R6.0.1 using H.323 signalling. Teldat devices provide autonomous Telephony Server Services to ensure telephone continuity even if there is no connectivity with the central server such as a WAN failure to Avaya Aura[®] Communication Manager.

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 compliance tested configuration between the Teldat VyDa Media Gateway and Avaya Aura[®] Communication Manager R6.0.1. The Teldat Media Gateway family, Teldat VyDa 1/2/3/4M VoIP Gateway (VyDa-M), offers a solution for each environment through modular platforms; the Teldat devices provide autonomous Telephony Server Services to ensure telephone continuity even if there is no connectivity with the central server such as a WAN failure to Avaya Aura[®] Communication Manager. Teldat has a wide range of cards and interfaces for telephone network connection (FXS/FXO, basic and primary ISDN and E&M). In this compliance test both FXO/FXS and E1 cards are connected to the VyDa-M for PSTN access in and out of the gateway.

The Avaya 96xx H.323 deskphones (Avaya deskphones) are registered to Avaya Aura[®] Communication Manager R6.0.1 (Communication Manager). The VyDa-M communicates to the Avaya deskphones via H.323 trunks between the VyDa-M and Avaya Aura[®] Communication Manager. A number of tests are carried out to ensure that in all cases calls are passed between the VyDa-M and the Avaya deskphones. Calls from the PSTN pass into the VyDa-M and then route to Avaya Aura[®] Communication Manager where the dial plan analysis in Avaya Aura[®] Communication Manager determines the endpoint to where the call needs to go.

2. General Test Approach and Test Results

The test environment comprises of three locations the main site and two remote sites 1 & 2. Each remote site has a VyDa-M gateway with a connection to the main site provided by H.323 Trunks between the VyDa-M and Communication Manager on the main site. Each VyDa-M can accommodate various PSTN connections through PCI slots on the VyDa-M. FXO/FXS and E1 PSTN connections were provisioned in these Application Notes. The test approach was to validate the correct operation of calls to and from the PSTN to the Avaya deskphones on each remote site. Also to validate the correct operation of calls between all three sites using H.323 trunks between each remote site and the main site. The tests were all functional in nature and performance testing was not included. All test cases passed successfully.

2.1. Compliance Test Cases

This section contains a summary of test cases carried out to validate the various capabilities of the solution and to show the interoperability of the VyDa-M gateways and Communication Manager.

2.1.1. PSTN calls to/from the Teldat VyDa 1/2/3/4M Gateway

This section shows the tests performed for the testing of calls to and from the PSTN to Avaya deskphones on each remote site using various PSTN connections on each VyDa-M and H.323 trunks to connect each remote site to the main site.

1. Call routing for incoming calls in the FXO/E1 lines (PSTN) of the VyDa-M to Communication Manager by H.323 signalling with destination an Avaya H.323 deskphone.
 - Incoming call from PSTN line in VyDa-M. VyDa-M initiates H.323 signalling to the Communication Manager over the H.323 trunks.
 - Communication Manager locates the Avaya H.323 deskphone/s that should receive the call and it sends the call to the correct Avaya H.323 deskphone.
 - Call is established and RTP Audio is flowing locally between Avaya H.323 deskphone and the PSTN caller.
2. Call routing for Outgoing Calls from Avaya H.323 deskphones to PSTN through VyDa-M.
 - Avaya H.323 deskphone sends call via H.323 trunk to VyDa-M.
 - Call is received in VyDa-M and it locates a free channel in PSTN voice ports to complete the call out.
 - Call is established between these two endpoints Avaya H.323 deskphone and the PSTN called set. RTP Audio is flowing locally between them.

3. Call routing for Incoming Calls through the PSTN of the VyDa-M to Communication Manager by H.323 signalling with destination FXS analog telephone in VyDa-M.
 - Incoming call from PSTN line in VyDa-M. VyDa-M initiates H.323 signalling to Communication Manager.
 - Communication Manager has the FXS extension in the dial plan to route to the VyDa-M.
 - VyDa-M is programmed to send the incoming call to the FXS port.
 - Call is presented to the FXS port of the VyDa-M and RTP Audio is flowing locally between PSTN caller and FXS ports of the VyDa-M.
4. Verify Call transfer between Avaya H.323 deskphones and VyDa-M FXS extension.
 - Make an incoming call over the PSTN into an Avaya H.323 deskphone. Answer the call, and transfer it to a VyDa-M FXS extension.
 - Transfer is completed and RTP flows between the PSTN caller and VyDa-M FXS extensions to which the call was transferred.
5. Verify group call pickup in an incoming call to an Avaya H.323 deskphone from a VyDa-M FXS extension.
 - Make an incoming call over the PSTN to an Avaya H.323 deskphone extension, then pick up the call from a VyDa-M FXS extension.
 - Call is attended by the VyDa-M FXS extension and RTP flows between the PSTN caller and VyDa-M FXS extension.
6. Verify call directed pickup in an incoming call to an Avaya H.323 deskphone from a VyDa-M FXS extension.
 - Make an incoming call over the PSTN into an Avaya H.323 deskphone then, send a directed pickup with extension number of the Avaya H.323 deskphone from the VyDa-M FXS extension.
 - Call is attended by the VyDa-M FXS extension and RTP flows between the PSTN caller and VyDa-M FXS extension.
7. Verify call forwarding for an incoming call to a VyDa-M FXS extension (always-on, no answer) from PSTN.
 - Make an incoming call over the PSTN into VyDa-M FXS extension then, the extension will forward the call to another extension (FXS or Avaya 96xx phone).
 - Call Forwarding works from VyDa-M FXS to any destination in two modes: “always-on” and “no answer” by the VyDa-M FXS extension and RTP flows between the PSTN caller and VyDa-M FXS extension that received the call forward.
8. Verify call conference of an incoming call to an Avaya H.323 deskphone to a FXS VyDa-M extension.
 - Make an incoming call over the PSTN into Avaya H.323 deskphone, then execute a conference to a FXS VyDa-M extension.
 - Call Conference works between an Avaya H.323 deskphone and a VyDa-M FXS extension and the PSTN caller and RTP audio flows correctly for the three attendees.

2.2. Test Results

All tests passed successfully.

Note: The Telephony Configuration on the VyDa-M determines what endpoints receive the call from the FXO ports. This setup is shown in **Section 6.1.7.1**.

2.3. Support

Support from Avaya is available at <http://support.avaya.com>. Technical support for the Teldat VyDa 1/2/3/4M Gateways can be obtained as shown below.

Teldat

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International Tel.: +34 918 076 630

Website: <http://www.teldat.com>

3. Reference Configuration

Figure 1 shows the network topology for compliance testing. The Teldat VyDa-M Gateway offers a VoIP connection to Communication Manager using H.323 Signalling over H.323 Trunks and a PSTN connection to the central office.

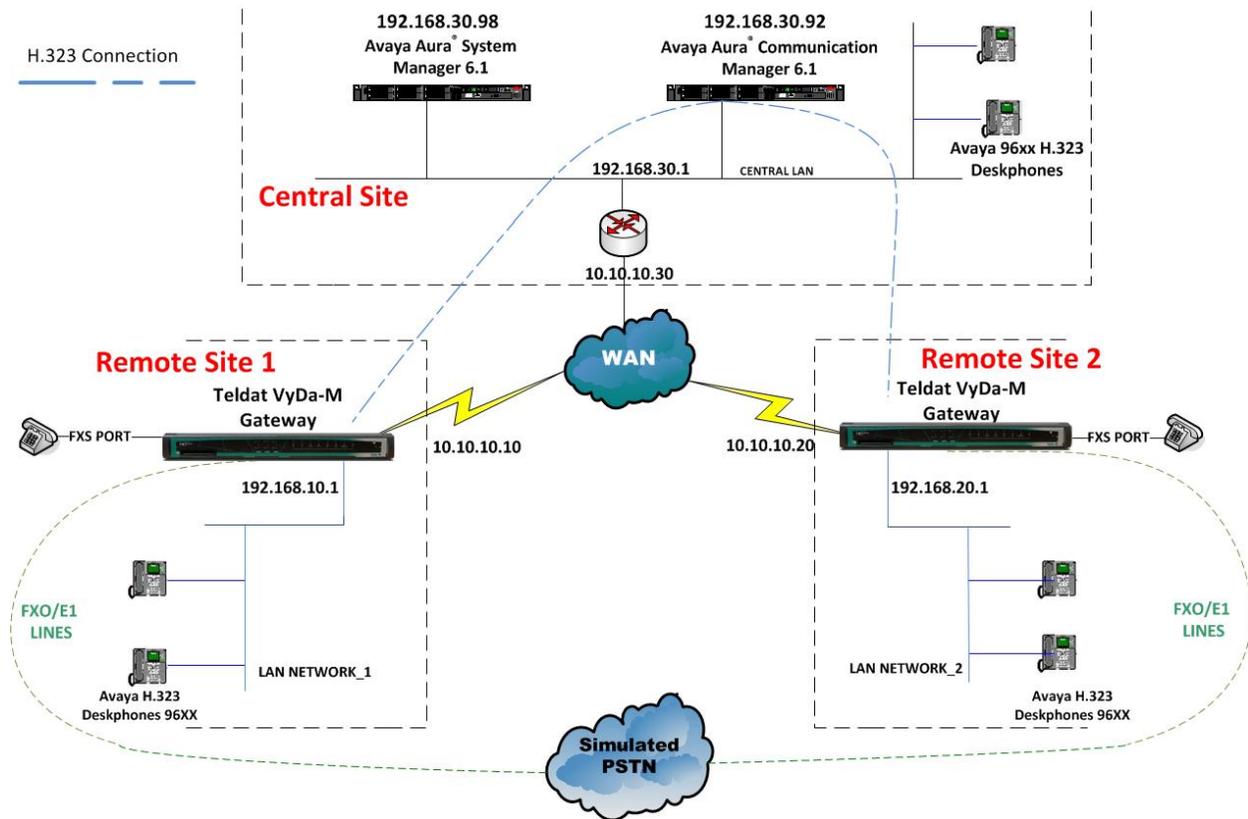


Figure 1: Connection of Teldat VyDa-1/2/3/4M Gateways and Avaya Aura[®] Communication Manager R6.0.1 using H.323 trunks

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment Description	Software Release
Avaya S8800 Server	Avaya Aura [®] Communication Manager R6.1 SP3
Avaya S8800 Server	Avaya Aura [®] System Manager R6.1 SP4
Avaya 9620 H.323 Sets	96xx H.323 Release 3.1 SP2
Avaya 9630 H.323 Sets	96xx H.323 Release 3.1 SP2
Teldat VyDa 1M Gateway	CIT Software release 10.7.54
Teldat VyDa 2M Gateway	CIT Software release 10.7.54
Teldat VyDa 3M Gateway	CIT Software release 10.7.54
Teldat VyDa 4M Gateway	CIT Software release 10.7.54

5. Configuration of Avaya Aura® Communication Manager

Configuration of the Communication Manager is performed through Avaya Site Administration supporting System Administration Terminal (SAT). An SSH session to the IP address of the Communication Manager initiates the console connection.

5.1. Create/Change Dial Plan

It is assumed that an existing dial plan is already in place. Type **change dialplan analysis** to make changes to the existing dial plan. In the example below **3005**, **4005**, **5001**, and **65xx** are associated with VyDa-M extensions and are therefore routed using automatic alternate routing (**aar**) to route calls to various endpoints when dialed.

```
change dialplan analysis                                     Page 1 of 12
                                                           DIAL PLAN ANALYSIS TABLE
                                                           Location: all                                     Percent Full: 2
Dialed Total Call      Dialed Total Call      Dialed Total Call
String Length Type    String Length Type    String Length Type
2          4   ext
3005       4   aar
4005       4   aar
5001       4   aar
65         4   aar
*          3   dac
```

5.2. Checking the Class of Restriction and System Parameters Features

The Class of Restriction (COR) determines what features the Communication Manager user associated with this COR can gain access to. Type **change cor 1** to make changes to class of restriction set 1. Ensure that **Can Be Picked Up By Directed Call Pickup** and **Can Use Directed Call Pickup** are both set to **y**.

```

change cor 1                                     Page 1 of 23
                                     CLASS OF RESTRICTION

COR Number: 1
COR Description: MainSite Class of Restriction

FRL: 0                                           APLT? y
Can Be Service Observed? n                     Calling Party Restriction: none
Can Be A Service Observer? n                   Called Party Restriction: none
Time of Day Chart: 1                           Forced Entry of Account Codes? n
Priority Queuing? n                             Direct Agent Calling? n
Restriction Override: none                     Facility Access Trunk Test? n
Restricted Call List? n                        Can Change Coverage? n

Access to MCT? y                               Fully Restricted Service? n
Group II Category For MFC: 7                   Hear VDN of Origin Annc.? n
Send ANI for MFE? n                           Add/Remove Agent Skills? n
MF ANI Prefix:                                Automatic Charge Display? n
Hear System Music on Hold? y PASTE (Display PBX Data on Phone)? n
Can Be Picked Up By Directed Call Pickup? y
Can Use Directed Call Pickup? y
Group Controlled Restriction: inactive
  
```

Type **change system-parameters features** to make changes to Feature Related System Parameters. Enable **Call Pickup Alerting** and **Directed Call Pickup** on **Page 19** of **system-parameters features**. Both features should be set to **y** as shown below.

```

change system-parameters features               Page 19 of 19
                                     FEATURE-RELATED SYSTEM PARAMETERS

IP PARAMETERS

Direct IP-IP Audio Connections? y
IP Audio Hairpinning? n
Synchronization over IP? n

CALL PICKUP
Maximum Number of Digits for Directed Group Call Pickup: 4
Call Pickup on Intercom Calls? y               Call Pickup Alerting? y
Temporary Bridged Appearance on Call Pickup? y Directed Call Pickup? y
Extended Group Call Pickup: none
Enhanced Call Pickup Alerting? y
Enhanced Call Pickup Delay Timer (sec.) Display: 5 Audible Notification: 5

Display Information With Bridged Call? n
Keep Bridged Information on Multiline Displays During Calls? y
PIN Checking for Private Calls? n
  
```

5.3. Adding H.323 Users

Type **add station xxxx** (where **xxxx** is the extension number of the deskphone to be added). Ensure that **COR** is set to **1** to use the COR changed in **Section 5.2**.

```
add station 3010                                     Page 1 of 5
                                                    STATION
Extension: 3010                                     Lock Messages? n          BCC: 0
Type: 9630                                         Security Code: *         TN: 1
Port: S00022                                       Coverage Path 1:        COR: 1
Name: Remotel,3010                                   Coverage Path 2:        COS: 1
                                                    Hunt-to Station:
STATION OPTIONS
Loss Group: 19                                     Time of Day Lock Table:
Speakerphone: 2-way                               Personalized Ringing Pattern: 1
Display Language: english                         Message Lamp Ext: 3010
Survivable GK Node Name:                          Mute Button Enabled? y
Survivable COR: internal                           Button Modules: 0
Survivable Trunk Dest? y                           Media Complex Ext:
                                                    IP SoftPhone? n
                                                    IP Video? n
Short/Prefixed Registration Allowed: default
                                                    Customizable Labels? Y
```

5.4. Adding Node Names

Each remote site is added to the list of node names in order to give an IP address as a destination to route the calls. Type **change nodes-names ip** in order to add the remote sites. Note below **Remotel** and **Remote2** were added as the two remote sites.

```
change node-names ip
                                                    IP NODE NAMES
Name          IP Address
CLAN          192.168.30.80
Medpro        192.168.30.81
Remotel     192.168.10.1
Remote2     192.168.20.1
SessionManager 192.168.30.101
default       0.0.0.0
gateway       192.168.30.1
procr         192.168.30.92
procr6       ::
```

5.5. Creating Trunk and Signaling Groups

A trunk group is created for each remote site in order to connect the sites to the main site. Type **add trunk-group x**, where x is the trunk group number, to create a new trunk group. The **Group Type** is **isdn** and the **Carrier Medium** is **H.323**. The **Signaling Group** number is the signaling group created below for this **trunk-group** in this example the value is **2**. The **Number of Members** will depend what Communication Manager is licensed for.

```
add trunk-group 2                                     Page 1 of 21
                                                    TRUNK GROUP

Group Number: 2                                     Group Type: isdn          CDR Reports: y
  Group Name: Remotel                               COR: 1                   TN: 1          TAC: *12
  Direction: two-way                               Outgoing Display? n     Carrier Medium: H.323
  Dial Access? y                                   Busy Threshold: 255    Night Service:
Queue Length: 0
Service Type: tie                                  Auth Code? n
                                                    Member Assignment Method: auto
                                                    Signaling Group: 2
                                                    Number of Members: 10
```

Create a signalling group for each trunk group created by typing **add signaling-group**. Note the following when creating the signalling group.

- **Group Number** **2**, the trunk group number created above
- **Group Type** **h.323**, for H.323 signalling
- **Near-end Node Name** **procr**, the name for the originating node. Note this can be CLAN also as shown in **Section 5.4**
- **Far-end Node Name** **Remotel**, the name for the endpoint node created in **Section 5.4**

```
add signaling-group 2                               Page 1 of 6
                                                    SIGNALING GROUP

Group Number: 2                                     Group Type: h.323
  SBS? n                                           Remote Office? n        Max number of NCA TSC: 0
  Q-SIP? n                                         Max number of CA TSC: 0
  IP Video? n                                       Trunk Group for NCA TSC:
  Trunk Group for Channel Selection: 2             X-Mobility/Wireless Type: NONE
  TSC Supplementary Service Protocol: a           Network Call Transfer? n
                                                    T303 Timer(sec): 10

H.245 DTMF Signal Tone Duration(msec):
Near-end Listen Port: 1720                         Near-end Node Name: procr
                                                    Far-end Node Name: Remotel
                                                    Far-end Listen Port: 1720
                                                    Far-end Network Region: 1
                                                    Calls Share IP Signaling Connection? n

  LRQ Required? n                                   Bypass If IP Threshold Exceeded? n
  RRQ Required? n                                   H.235 Annex H Required? n
                                                    Direct IP-IP Audio Connections? y
  DTMF over IP: out-of-band                       IP Audio Hairpinning? n
  Link Loss Delay Timer(sec): 90                   Interworking Message: PROGRESS
  Enable Layer 3 Test? n                           H.323 Station Outgoing Direct Media? n
                                                    DCP/Analog Bearer Capability: 3.1kHz
```

5.6. Creating a Route Pattern and Automatic Alternate Routing

A route pattern is created to point the dialled digits to a particular trunk group. Type **change route-pattern x**, where x is the route-pattern value. Note the group number (**Grp No**) is the trunk group created in **Section 5.5**. A unique name is added for the **Pattern Name**.

Note: For compliance testing two route-patterns 2 & 3 are created sending calls to **Grp No**'s (group numbers) 2 & 3. Calls using route-pattern 2 are sent to remote site 1 and calls using route-pattern 3 are sent to remote site 2.

```

change route-pattern 2                                     Page 1 of 3
                Pattern Number: 2   Pattern Name: remote 1
                SCCAN? n           Secure SIP? n
Grp No  FRL NPA Pfx Hop Toll No.  Inserted          DCS/ IXC
                Mrk Lmt List Del  Digits          QSIG
                Dgts
1: 2      0
2:
3:
4:
5:
6:

                BCC VALUE  TSC CA-TSC   ITC BCIE Service/Feature PARM No. Numbering LAR
                0 1 2 M 4 W      Request          Dgts Format
                Subaddress
1: y y y y y n  n           unre          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

```

Create automatic alternate routing entry (AAR) by typing **change aar analysis**. Note the entries for **3005**, **4005**, **5001** and **65xx** are using various route patterns created above.

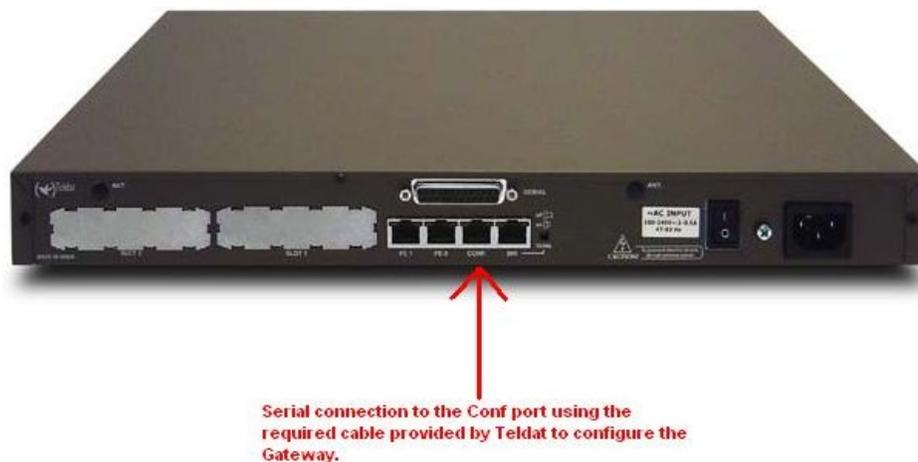
```

change aar analysis 1                                     Page 1 of 2
                AAR DIGIT ANALYSIS TABLE
                Location: all           Percent Full: 1
                Dialed      Total      Route      Call      Node      ANI
                String      Min Max  Pattern  Type      Num      Reqd
                2           7 7    999     aar       n        n
3005           4 4  2     aar       n
4005           4 4  3     aar       n
5001           4 4  2     aar       n
65            4 4  3     aar       n
                7           7 7    999     aar       n        n
                8           7 7    999     aar       n        n
                9           7 7    999     aar       n        n

```

6. Configuration of the Teldat VyDa-M Gateway

The VyDa 1M-2M-4M gateways have an RJ45 connector on the rear or front panel referred to as **CONF** which provides access to the device's local console. The example below shows the connection on a VyDa 2M. Connect the CONF port to a PC with terminal emulation such as putty, hyperterminal, SecureCRT, TeraTerm or other terminal emulator VT100. The VyDa 3M gateway has a DB9 pin connector in the rear panel referred to as **AUX** which provides access to the device's local console.



Configuration setup for the terminal connection must be:

- Speed: 9600 bps
- Eight data bits
- One stop bit
- No parity bit
- No type of flow control

Once connected the following console screen and symbol appears.

```
Teldat                (c)2001-2009
Router model VyDa-2M 7 148 CPU MPC8260      S/N: 106/13435
2 LAN
CIT software version: 10.7.54
*
```

In order to proceed with configuration **process 4** is entered at the * prompt. The configuration, in text mode, can be copied and pasted into the console from the prompt **Config>**

```
Teladat                (c) 2001-2009

Router model VyDa-2M 7 148 CPU MPC8260      S/N: 106/13435
2 LAN
CIT software version: 10.7.54

*process 4
Config>
```

6.1. Showing Menus and Submenus Configuration for VyDa-M Gateway

This section will go through each part of the VyDa-M setup and configuration. As explained above the total configuration can be written and pasted into the console in one go however in order to show the various commands necessary for each PSTN connection and H.323 setup a subset of these commands are shown below. All commands are entered through a serial connection to the VyDa-M as shown in the beginning of this section. A terminal program such as Reflections or Hyperterminal can be used to issue the commands.

6.1.1. Common Configuration Commands

The following commands are entered common to all VyDa-M configurations in order to show command errors.

- Setup the VyDa to log errors so as any issues can be reported by using the command **log-command-errors**.
- Set the **hostname** of the device for reference in logs etc.
- The TNIP (GRE Tunnel) is virtually added by the command **add device tnip <n>**. This GRE tunnel is used to interconnect the LAN to central LAN. This interface is a virtual entity in the gateway for only routing purposes and is added in the global configuration.

```
log-command-errors
no configuration
set hostname 2M_REMOTE_SITE_2
add device tnip 1
```

6.1.2. IP Configuration Commands

In section **network** enter the IP address of the LAN (**ethernet 0/1**) and WAN (**ethernet 0/0**) for the VyDa-M gateway. In the example below the internal LAN IP address is **192.168.20.1** and the External WAN address is **10.10.10.20**.

```
*process 4
Config> network ethernet0/0
-- Ethernet Interface User Configuration --
ethernet0/0 config>
    ip address 10.10.10.20 255.255.255.0
    exit
;
Config> network ethernet0/1
-- Ethernet Interface User Configuration --
ethernet0/1 config>
    ip address 192.168.20.1 255.255.255.0
    exit
Config>
;
```

Add a GRE tunnel (WAN to WAN) in order to get direct connectivity between both LAN networks.

```
Config> network tnipl
-- IP Tunnel Net Configuration --
tnipl config>
description "GRE TUNNEL TO CENTRAL"
;
    ip address 2.2.2.2 255.255.255.0
;
    enable
    mode gre ip
    source 10.10.10.20
    destination 10.10.10.30
    keepalive 10 3
    exit
Config>
;
```

In section **protocol ip** configure the IP routing and DNS-domain as follows. Type **protocol ip** to enter the IP configuration.

```
Config> protocol ip
-- Internet protocol user configuration --
IP config>
; -- Internet protocol user configuration --
    internal-ip-address 192.168.20.1
    route 0.0.0.0 0.0.0.0 2.2.2.1
    dns-domain-name avaya.com
    classless
exit
;
Config>
;
```

6.1.3. VoIP/PSTN Interface Configuration Commands

VyDa-M gateways support different VoIP plug-in cards in order to connect to the PSTN.

- FXS/FXO card
- E1 ISDN card

The VoIP cards are automatically recognized & installed by the VyDa-M CIT software. Once the VyDa-M gateway has been powered up the following configuration will appear by default.

FXS/FXO card Default Configuration.

```
network voip1/0
-- VoIP interface Configuration --
  line 1 interface-type fxs
  line 2 interface-type fxs
exit
```

E1 ISDN card Default Configuration.

```
network g703port2/0
; -- T1-E1 / G703 Configuration --
;
  exit
;
network voip2/0
; -- VoIP interface Configuration --
  base-interface
; -- Base Interface Configuration --
  base-interface g703port2/0 255 link
;
  exit
;
  exit
;
```

In order to fully configure the VoIP interfaces the exact configuration of each PSTN line, FXO or E1 must be added. The following examples show the setup used in the compliance testing for each line.

6.1.3.1 FXS/FXO VoIP Interface Configuration Commands

The FXS/FXO card on the VyDa-M Gateway will translate from analog to VoIP (H.323) using an IP codec of either G.729 or G.711. Each line of the FXS/FXO card is configurable as FXS or FXO depending on the connected device. The FXS is used to connect analog telephones and FXO is used to connect PSTN analog lines. In the configuration below an analog set was configured on **line 1** with extension number 4005 (**fxs**) and an analog line added on **line 2** with telephone number 5001(**fxo**). For the FXO line the following configuration line is added **line 2 direct-dialing 1 0s** in order for a dial-tone detected the VyDa-M will generate an internal call to the destination number 1 in 0 seconds. This internal call will be established with the Telephony options in **Section 6.1.7**.

```
Config>
;
network voip1/0
-- VoIP interface Configuration --
line 1 interface-type fxs
line 1 telephone-number 4005
;
line 2 interface-type fxo
line 2 direct-dialing 1 0s
line 2 telephone-number 5001
dsp-firmware g729-g711
exit
Config>
;
```

6.1.3.2 E1 ISDN VoIP Interface Configuration Commands

This configuration includes.

- Global-profiles dial
- G703 Interface configuration parameters
- VoIP E1 configuration parameters

The **global-profiles dial** configuration section defines the behavior of the G703 interface to accept incoming and outgoing calls, and to define the interface as audio type for ISDN signaling.

The **G703 configuration** is used to define the physical E1 line parameters. Note that each of these parameters are set depending on the PSTN connection and the public exchange setup.

- **Framing** (CRC4 or non-CRC4)
- **Clocksource** (Internal or External)
- **Emulate** (Emulation as Network or User)
- **PRI-group** (Number of timeslots used of the E1 circuit)

The **VoIP interface configuration** is used to associate the dial profile and the number of circuits related to that voice over IP interface. Then this VoIP interface will be used in Telephony menu in order to associate dial-peers for calling match process.

```
Config>
;
  global-profiles dial
; -- Dial Profiles Configuration --
  profile VoIP_Audio default
  profile VoIP_Audio inout
  profile VoIP_Audio isdn-type audio
  exit
;
  network g703port2/0
; -- T1E1 / G703 Configuration --
  framing no-crc4
  clocksource internal
  emulate network
  isdn overlap-dial
  pri-group timeslots 1-3,16
  exit
;
  network voip2/0
; -- VoIP interface Configuration --
  base-interface
; -- Base Interface Configuration --
  base-interface g703port2/0 255 link
  base-interface g703port2/0 255 profile VoIP_Audio
  base-interface g703port2/0 255 number-of-circuits all
;
  exit
;
  exit
;
Config>
```

6.1.4. Network Service Monitor Configuration

The Network Service Monitor (NSM) operation 1 is sending an ICMP packet to the specified destination IP address (Communication Manager) each 3 seconds from the specified **source-ipaddress**. This NSM operation is responsible to detect the Communication Manager availability at IP level as each ICMP response is expected from the destination IP address that the NSM operation is supervising. Type **feature nsm** to enter the **Network Service Monitor Configuration**.

```
Config> feature nsm

; -- Network Service Monitor configuration --
  operation 1
; -- NSM Operation configuration --
  type echo ipicmp 192.168.30.92
  frequency 3
  source-ipaddr 192.168.20.1
  exit
;
  schedule 1 life forever
  schedule 1 start-time now
  exit
;
Config>
;
```

6.1.5. Network Service Level Advisor Configuration

The Network Service Level Advisor (NSLA) is defined in order to detect an IP level communication disruption with the Communication Manager. This is an example with a WAN ethernet interface with has a lower round trip time (rtt) than 2ms. The rtt sensibility thresholds are set to the limits in the WAN access. There are two advisors created: 1 and 2. Type **feature nsula** to enter the **Network Service level Advisor**.

- **Advisor 1** will keep the state as ACTIVE by default
- **Advisor 2** will keep the state as DEACTIVE by default

```
Config> feature nsula

; -- Feature Network Service Level Advisor --
  enable
  filter 1 nsm-op 1 rtt
  filter 1 significant-samples 2
  filter 1 activation threshold 300
  filter 1 activation sensibility 50
  filter 1 activation stabilization-time 1
  filter 1 deactivation threshold 300
  filter 1 deactivation sensibility 100
  filter 1 deactivation stabilization-time 1
;
  alarm 1 filter-id 1
  advisor 1 not alarm-id 1
  advisor 2 alarm-id 1
  exit
;
Config>
```

6.1.6. H.323 Protocol Configuration Commands

Enter the information about the H.323 protocol in the VyDa-M gateway by typing **protocol h.323** at the **Config>** prompt.

Parameters to be configured:

- **application address (192.168.20.1)** is the IP address that this protocol will use to send H.323 signaling to any device.
- **application gateway** defines the translation from analog to H.323 signaling.
- **gatekeeper address (192.168.30.92)** issues the H.323 gatekeeper IP address in order to check registration status of the VyDa-M gatekeeper.
- **gw name Teldat-VYDA** issues the VyDa device's name in the H.323 gatekeeper.
- **h245 capability all-dtmf** issues the DTMF relay negotiation with all the existing possibilities: RFC2833, alphanumeric and signal in the H.323 signaling process.
- **no fast-connect** disables the fast connection method for H.323 signaling protocol.

```
Teldat                (c)2001-2009

Router model VyDa-2M 7 148 CPU MPC8260      S/N: 106/13435
2 LAN
CIT software version: 10.7.54

*process 4
Config>protocol h.323

    application address 192.168.20.1
    application gateway
    gatekeeper address 192.168.30.92
    gw name Teldat-VYDA
    h245 capability all-dtmf
    no fast-connect

exit
Config>
```

6.1.7. Telephony Configuration Commands

The telephony menu allow us to introduce the incoming/outgoing calls behavior and other features such as: group calls pickup, directed calls pickup, call forward, etc... Enter the information about the telephony in the VyDa-M gateway by typing **telephony** at the **Config>** prompt. Under **telephony** menu, a voice-class that includes a list of codec's that will be used to negotiate on each H.323 signaling is defined.

```
Config>telephony
;
; -- Telephony configuration --
    voice-class 15
        codec-preference g711alaw
        codec-preference g711ulaw
        codec-preference g729
    exit
Config>
```

6.1.7.1 Telephony Configuration with FXO LINE

Dial-peers are created which are virtual call routing rules for different VoIP ports. The following configuration is used for an FXO LINE as the PSTN line that provides voice services to the public voice network.

The **dial-peers** 1 and 2 are **H323** dial-peers that have a dynamic target. Dynamic target means a local H.323 registered extension will only match if there are H.323 extensions registered to the local H.323 gatekeeper. The dial-peer 21 and 22 are group dial-peers that will send H.323 signaling to the peer-groups 1 and 2 respectively when the Communication Manager H.323 endpoint is UP or DOWN. In this example, the configuration for the H.323 set was defined with extension number: 4010 and FXS extensions: 4005 and 4006.

The dial-peers 23, 24 and 25 are voice-ports dial-peers that have VoIP ports as target. Dial-peers 23 and 24 have FXS analog telephones as target. Dial-peer 25 has FXO line as target for incoming / outgoing calls.

The dial-peer 20 is a **H323** dial-peer that has an H.323 endpoint as the target of the H.323 signaling. This peer was used to send all calls to Communication Manager when the link between the VyDa-M and Communication Manager is up.

```

Config>telephony
;
; -- Telephony configuration --
  dial-peer 1 H323
    no vad
    destination-pattern 4000
    dtmf-relay h245-signal
    target dynamic
    track nsla-advisor 2
    voice-class 15
  exit
;
  dial-peer 2 H323
    no vad
    destination-pattern 4001
    dtmf-relay h245-signal
    target dynamic
    track nsla-advisor 2
    voice-class 15
  exit
;
  dial-peer 21 group
    no register h323
    destination-pattern 1
    target group 1
    track nsla-advisor 1
  exit
;
  dial-peer 22 group
    no register h323
    destination-pattern 1
    forward no-answer peer-group 5
    forward no-answer timeout 10s
    target group 2
    track nsla-advisor 2
  exit
;
  dial-peer 23 voice-port
    description "FXS PORT 1"
    no vad
    destination-pattern 4005
    target voice-port voip1/0 1
    voice-class 15
  exit
;
  dial-peer 24 voice-port
    description "FXS PORT 3"
    no vad
    destination-pattern 4006
    target voice-port voip1/0 3
    voice-class 15
  exit
;
  dial-peer 25 voice-port
    description EMERGENCY_OUTGOING_CALLS_TO_PSTN
    no vad
    destination-pattern .T
    destination-pattern 6510
    destination-pattern 6500
    destination-pattern 6501
    destination-pattern 6502

```

```

        destination-pattern 6503
        destination-pattern 6504
        destination-pattern 6511
        target voice-port voip1/0 2
        voice-class 15
    exit
;
    dial-peer 20 h323
        description "FROM PRI H323 TO AVAYA"
        destination-pattern 4000
        destination-pattern 4001
        destination-pattern 3...
        destination-pattern 2000
        destination-pattern 5000
        destination-pattern 4005
        destination-pattern 4006
        destination-pattern 4010
        destination-pattern 4011
        dtmf-relay h245-signal
        incoming dial-plan peer-group 3
        target ipv4 192.168.30.92
        track nsia-advisor 1
        voice-class 15
    exit
;
    peer-group 1
        dial-peer 20 4010
    exit
;
    peer-group 3
        dial-peer 23
        dial-peer 24
        dial-peer 25
    exit
;
    peer-group 4
        dial-peer 23 4005
        dial-peer 24 4006
        dynamic-peers
    exit
;
    peer-group 5
        dial-peer 1 4000
    exit
;
    peer-group 2
        dial-peer 23 4005
        dial-peer 24 4006
    exit
;
    exit
;

```

6.1.7.2 Telephony Configuration with E1 Q931

This configuration with E1 line using 3 channels as ISDN channels is different from FXO as the E1 line has normally a different DDI number for each line. In this example the numbering plan for the E1 is 65xx and the **dial-peer 20 h323** is translating it to local extensions with numbering plan: 30xx. Then it's sent to the H.323 endpoint.

```
telephony
; -- Telephony configuration --
  dial-peer 20 h323
    description "FROM PRI-R2 H323 TO AVAYA"
    destination-pattern 65..
    destination-pattern 6...
    destination-pattern 3...
    dtmf-relay h245-signal
    huntstop busy
    incoming called number T
    incoming dial-plan peer-group 3
    outgoing prefix 30
    outgoing strip-digits 2
    target ipv4 192.168.30.92
    track nsla-advisor 1
    voice-class 15
  exit
;
  dial-peer 22 voice-port
    description PRI-ISDN-CS1K-PSTN
    no vad
    destination-pattern .T
    target voice-port voip2/0 1
    voice-class 15
  exit
;
  dial-peer 23 voice-port
    description "FXS PORT 1"
    no vad
    destination-pattern 3005
    destination-pattern 6505
    forward no-answer peer-group 5
    forward no-answer timeout 10s
    target voice-port voip1/0 1
    voice-class 15
  exit
;
  dial-peer 24 h323
    no vad
    destination-pattern 65..
    dtmf-relay h245-signal
    outgoing prefix 30
    outgoing strip-digits 2
    target dynamic
    track nsla-advisor 2
    voice-class 15
  exit
;
  dial-peer 30 facility
    destination-pattern *8
    target group-pickup peer-group 4
  exit
```

```

;
dial-peer 31 facility
  destination-pattern *9T
  target directed-pickup prefix 2
exit
;
dial-peer 25 h323
  no vad
  destination-pattern 30..
  dtmf-relay h245-signal
  target dynamic
  track nsla-advisor 2
  voice-class 15
exit
;
;
peer-group 3
  dial-peer 23
exit
;
peer-group 4
  dial-peer 23 3005
  dynamic-peers
exit
;
peer-group 5
  dial-peer 20 3010
exit
;
exit
;

```

Once the whole configuration has been finished, save the configuration by typing **save yes** at the **Config>** prompt and restart the device.

```

Config>save yes
Building configuration as text... OK
Writing configuration... OK on Flash
Config>end
*restart
Are you sure to restart the system(Yes/No)? yes

```

7. Verification Steps

The following steps can be carried out to ensure the correct configuration was setup for both Communication Manager and VyDa-M.

7.1. Verification of Avaya H.323 deskphones routing calls to VyDa-M gateway

On Communication Manger there are traces that can be run to show the messages going across a trunk group. Type **display trunk group x** (where x is the number of the H.323 trunk group setup. Take note of the **TAC** entry on the page displayed as shown below.

```
display trunk-group 2                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 2                Group Type: isdn                CDR Reports: y
  Group Name: Remotel          COR: 1                    TN: 1          TAC: *12
  Direction: two-way          Outgoing Display? n        Carrier Medium: H.323
  Dial Access? y              Busy Threshold: 255       Night Service:
Queue Length: 0
Service Type: tie              Auth Code? n
                                     Member Assignment Method: auto
                                     Signaling Group: 2
                                     Number of Members: 10
```

The TAC is used to trace the H.323 messages on the trunk. Type **list trace tac *12** to show the H.323 messages for the Trunk group that has a tac entry of *12.

```
list trace tac *12                                       Page 1
                                     LIST TRACE
time          data
16:14:18 TRACE STARTED 09/22/2011 CM Release String cold-00.1.510.1-19009
```

H.323 messages can be traced on a per set basis also. Type **list trace station xxxx** (where xxxx is the H.323 extension that is to be traced).

```
list trace station 2012                                   Page 1
                                     LIST TRACE
time          data
16:15:50 TRACE STARTED 09/22/2011 CM Release String cold-00.1.510.1-19009
16:15:53 active station      2012 cid 0x1c9
16:16:00 G711A ss:off ps:20
          rgn:1 [192.168.30.55]:3116
          rgn:1 [192.168.30.81]:3216
16:16:03 dial 2010
16:16:03 ring station      2010 cid 0x1ca
16:16:03 G711A ss:off ps:20
```

7.2. Logs from Teldat VyDa-M to show FXO and E1 (PSTN connection) is up

The Teldat VyDa 1M/2M/3M/4M gateway has a monitoring menu that can be accessed through its console port. The monitoring menu allows the status check of each physical and virtual port and the statistics of the current working configuration of the VyDa-M. In order to access to the monitoring menu type **process 3** at the * prompt.

7.2.1. Interfaces Monitoring

Type **process 3** at the * prompt in order to enter monitoring. Type **conf** at the + prompt to get a view of all interfaces. Each PSTN interface such as FXO, E1, BRI can be seen as either up or down. Each Ethernet interface (LAN and WAN) can also be monitored as up or down.

```
Teldat (c)2001-2009

Router model VyDa-2M 7 148 CPU MPC8260 S/N: 106/13435
2 LAN
CIT software version: 10.7.54

*
*process 3
+
;
+conf

Teldat's Router, VYDA-2M 7 71 S/N: 106/04589
P.C.B.=8c Mask=0c10 Microcode=134f0 CLK=262144 KHz BUSCLK=65536 KHz PCICLK=32768
KHz
ID: AT150-16F128R L7.71

Boot ROM release:
BIOS CODE VERSION: 03.01 Mar 6 2007 18:02:45
gzip Jan 22 2007 19:11:47
io1 Jan 22 2007 19:11:24
io2 Jan 22 2007 19:11:24
io3 Mar 6 2007 18:02:38
START FROM FLASH L1 Watchdog timer Enabled

Software release: 10.7.54 Jul 23 2007 19:35:09
Compiled by INTEGRATOR on INTEGRATOR2000
Loaded from primary partition

Hostname: VYDA_PRI Active user: ccea
Date: Monday, 09/12/11 Time: 10:58:17
Router uptime: 1w6d15h31m49s

Num Name Protocol
0 IP DOD-IP
3 ARP Address Resolution Protocol
4 H323 H323
6 DHCP Dynamic Host Configuration Protocol
11 SNMP SNMP
13 RIP Route Information Protocol
17 SIP SIP
28 PPPoE Point-to-Point Protocol Over Ethernet
30 EAPOL Extensible Authentication Protocol Over LAN
```

31 Preauth WLAN Preauthentication

7 interfaces:

Connector	Interface	MAC/Data-Link	Status
FE0/LAN1	ethernet0/0	Ethernet/IEEE 802.3	Up
FE1/LAN2	ethernet0/1	Ethernet/IEEE 802.3	Down
BRI/ISDN1	bri0/0	BRI Net	Up
---	x25-node	internal	Up
SLOT1	voip1/0	internal	Up
SLOT1	g703port1/0	PRI/E1 Net	Up
---	tni1	internal	Testing

SNMP OperStatus:

Interface	OperStatus
ethernet0/0	Up
ethernet0/1	Down
bri0/0	Up
x25-node	Up
voip1/0	Up
g703port1/0	Up
tni1	Down

Encryption Engines:

Hardware: SEC-8272 Revision: 0xA

+

7.2.2. H.323 Protocol Monitoring

Type **process 3** at the * prompt to enter monitoring. At the + prompt in order to list the display prompts type **display ?** Type **display gatekeeper** to see the status of the H.323 gatekeeper, which should be **idle** as shown below.

```
Tel dat                      (c) 2001-2009

Router model VyDa-2M 7 148 CPU MPC8260      S/N: 106/13435
2 LAN
CIT software version: 10.7.54

*
*process 3
+
2M_REMOTE_SITE_1 H323 Mon+display ?
  cause          RAS release codes
  gatekeeper     View the application gatekeeper status
  ras           View RAS status and identifiers
2M_REMOTE_SITE_1 H323 Mon+display gatekeeper

App gatekeeper state: Idle state
```

Other areas of the VyDa-M can also be monitored. Type **telephony** at the + prompt to enter the telephony monitoring. Type **?** at the + prompt to list the commands used for further monitoring.

```
1M_REMOTE_SITE_1 H323 Mon+

1M_REMOTE_SITE_1 +telephony

Telephony Monitor

1M_REMOTE_SITE_1 Telephony Mon+?
  call-make      Carry out a call to a telephone number
  call-waiting   Put a call associated to an interface line on hold
  clear         Delete statistics
  display       Display the status of the voice messages
  dump-cdrs     Record cdrs to the configured file
  hang-up       Hang up a line that is not hook-on
  list         List system info
  no           Disable a monitoring command
  pick-up       Pick up a line that is hook-on
  srtp         SRTP monitoring commands
  trace        Debugging command for Teldats use
  voice-message Reproduce a previously recorded voice message
  exit
```

8. Conclusion

As illustrated in this Application Notes Teldat VyDa 1/2/3/4M Gateways can be configured to successfully interoperate with Avaya Aura® Communication Manager R6.0.1. All tests passed with no failures. All PSTN calls to and from the Teldat VyDa-M gateways completed successfully.

9. Additional References

This section references documentation relevant to these Application Notes. The Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Administering Avaya Aura® Communication Manager, Document ID 03-300509, June 2010.*
- [2] *Avaya Aura® Communication Manager Feature Description and Implementation, Document ID 555-245-205, June 2010.*
- [3] *Avaya one-X Deskphone SIP Administrator Guide, Release 6.1, Document ID 16-603838, December 2010.*

Product documentation on Teldat's VyDa 1/2/3/4 M Gateway is available at <http://www.teldat.com>

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