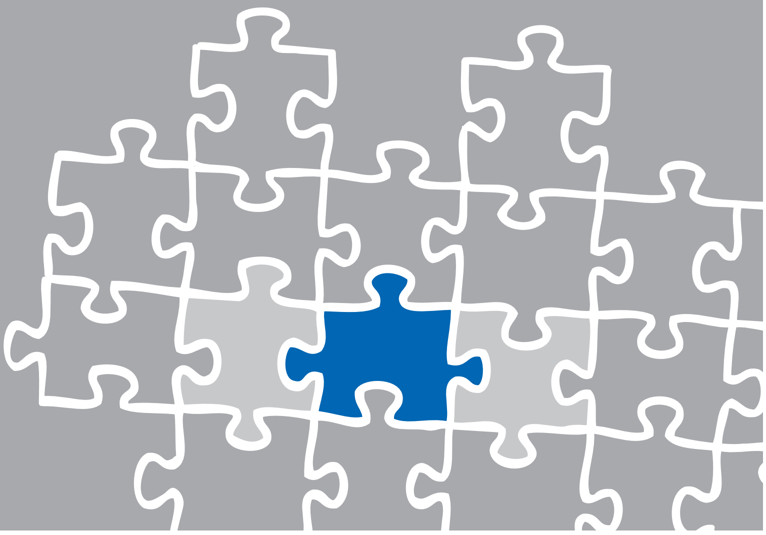
TechNote

Avaya Aura Communication Manager

November 18, 2015

XCAPI voiceoverIP



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Introduction

This document is intended to support engineers with the integration of the latest XCAPI version into an existing Avaya Aura Communication Manager environment. Though being based on version 7.0 of the Avaya Aura Communication Manager (using a g450 media gateway) and XCAPI version 3.5.59 this document is also applicable to other versions with a few adjustments.

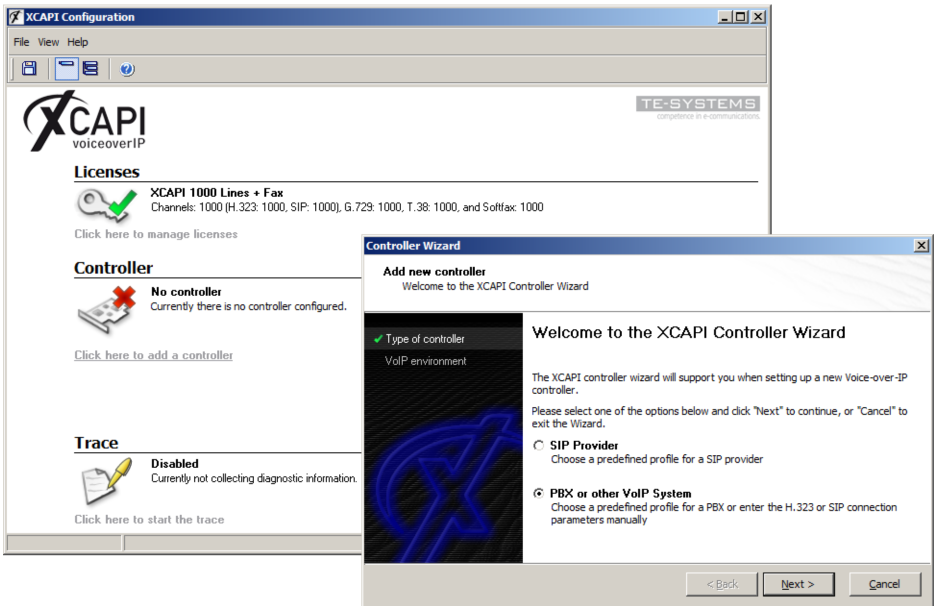
The following pages give essential information to allow optimal interworking of both the Avaya Aura Communication Manager and XCAPI. At this point we suppose that the Avaya Aura Communication Manager environment, the hardware and the operating system where XCAPI and the CAPI 2.0 application is running on, are properly installed and accessible through the IP network.

For detailed Avaya Aura Communication Manager configuration procedures, please refer to the respective manufacturer documentations and manuals. Additional XCAPI information and documents (TechNotes), e.g. Quick Starter Guide, License on demand, Fax Transmission, Virtual Hardware ID and VMware Virtual Machines can be found on our XCAPI Website within the community download section and on our YouTube channel.

XCAPI Configuration

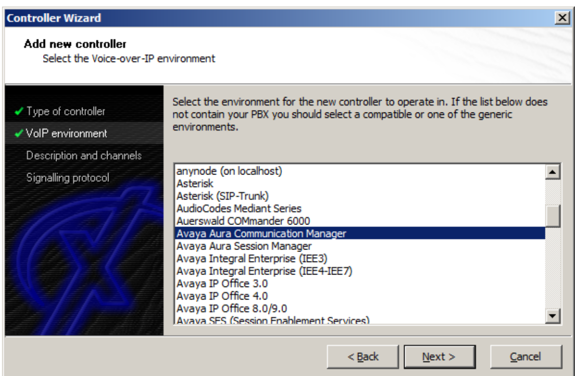
Please start up the XCAPI configuration to create a new controller that will be assigned to the Avaya Aura Communication Manager SIP trunk. The Avaya Aura Communication Manager SIP trunk configuration is described from page 8.
If you've just installed XCAPI and start the configuration tool for the first time, the XCAPI controller wizard will pop up automatically. This also happens if no controller is configured.

To start up the XCAPI controller wizard on your own, just click the hyperlink labeled **Click here to add a controller** on the main page of the XCAPI configuration. On the first controller wizard dialog, please select the **PBX or other VoIP System** and proceed with the **Next** button.



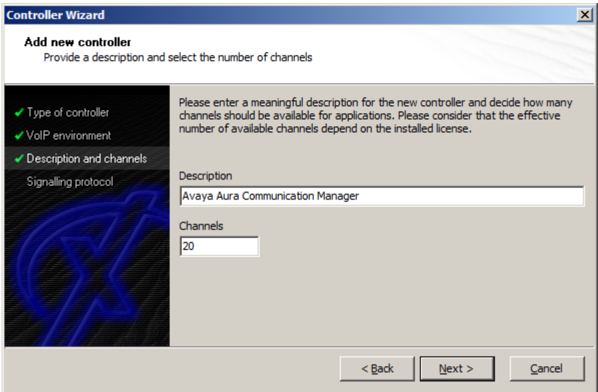
2.1 Voice over IP Environment

The next dialog of the configuration tool shows a list of some common Voice-over-IP environments. Selecting one of those will configure the XCAPI with a selection of near-optimal defaults for the kind of environment you have, saving you a lot of manual configurations.



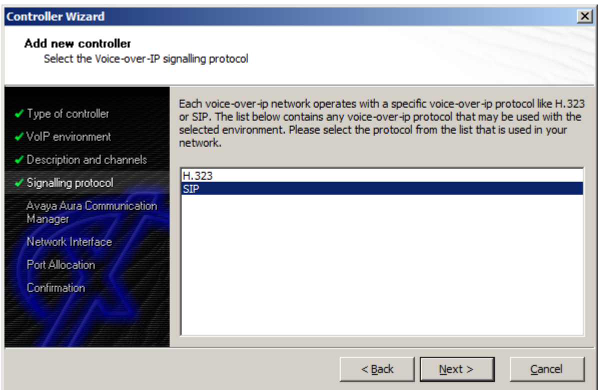
2.2 Description and Channels

This dialog allows you both to enter an appropriate controller name and set up the number of available and licensed channels. So please enter the amount of simultaneous channels XCAPI should provide when communicating with the Avaya Aura Communication Manager and the CAPI 2.0 application.



2.3 Signaling Protocol

Please select the appropriate signaling protocol used for this VoIP environment.

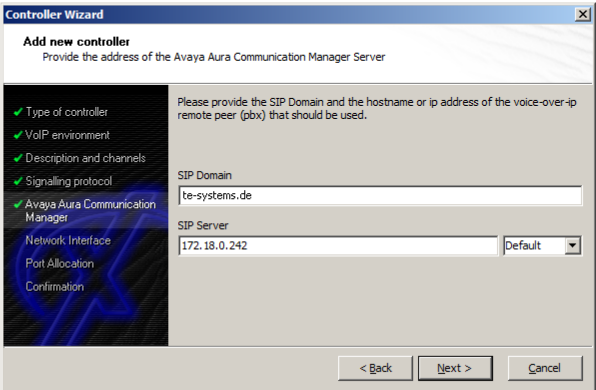


2.4 Gateway Address

Next, please provide the host name and/or the IP address of the SIP listening Avaya Aura Communication Manager Ethernet interface. Please note that both, the XCAPI controller and Avaya Aura Communication Manager use by default the TCP port 5060 for SIP signaling.

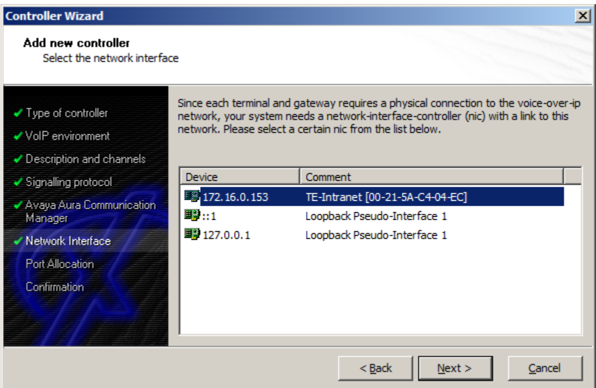
Enter the according SIP domain if a fully qualified domain name is used as **Far-end Domain** within the XCAPI related SIP signaling group.

Wrong domain handling will evoke a **403 Forbidden(Invalid domain in From: header)** failure.



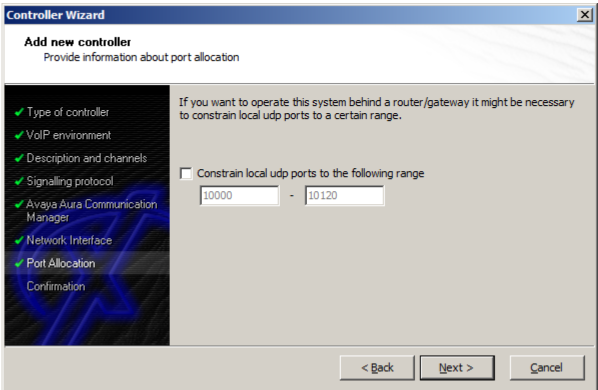
2.5 Network Interface

Select the network interface you want to connect to the newly created XCAPI controller.



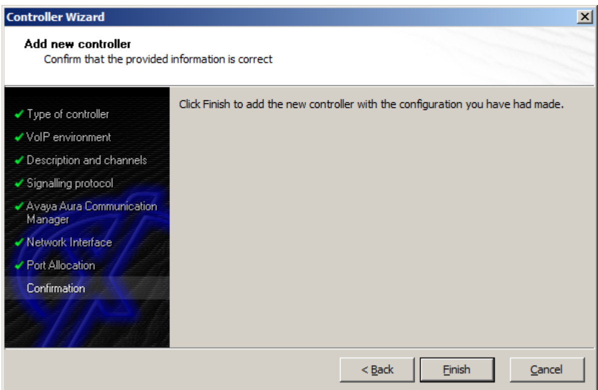
2.6 Port Allocation

On demand an UDP port range for RTP and T.38 can be specified. This range will be used by the XCAPI controller towards the gateway.

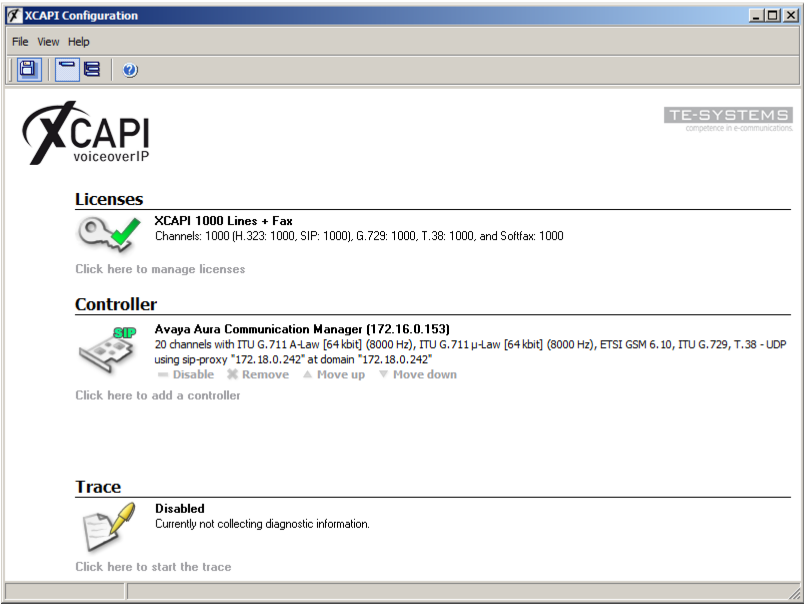


2.7 Confirmation

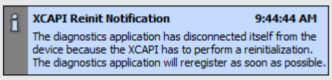
The final dialog of the controller wizard performs some checks on the configuration parameters you've made. When everything is correct, please use the **Finish** button in order to create the new controller.



Finally you can save the controller which is also listed on the main view of the XCAPI configuration.



The bound CAPI 2.0 application with its services must always be restarted to take effect on the XCAPI controller changes. Restarting any of the XCAPI services won't help at all.



Avaya Aura Communication Manager Configuration

In order to enable the communication between the Avaya gateway and XCAPI, the appropriate SIP trunk configuration must be provided. This chapter reviews the essential SIP trunk configuration, where the Communication Manager covers the typical gateway tasks such as VoIP trunking, codec settings and numbering analyzing. This configuration must of course be adjusted to your VoIP environment.

3.1 Licenses

Please review the license availability (Maximum Administered SIP Trunks) within the **system-parameters customer-options**.

```
display system-parameters customer-options                      Page   2 of   12
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 4000  30
           Maximum Concurrently Registered IP Stations: 2400  2
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 50    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  0
                   Maximum Video Capable IP Softphones: 50    0
                       Maximum Administered SIP Trunks: 4000  125
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    1

        (NOTE: You must logoff & login to effect the permission changes.)
```

3.2 System Parameters

For allowing trunk-to-trunk connections and call forwarding to remote locations, you have to ensure that the **Trunk-to-Trunk Transfer** parameter is set to **all**.

```
display system-parameters features                              Page   1 of   19
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? y
                                      Trunk-to-Trunk Transfer: all
              Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                  AAR/ARS Dial Tone Required? n

             Music (or Silence) on Transferred Trunk Calls? all
             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
```

3.3 Node Names IP

In this example, **XCAPI** is assigned to 172.16.0.153 which will be utilized in the signaling group, as described in the same named chapter starting on page 13.

```
display node-names ip                                          Page   1 of   2
                                    IP NODE NAMES
    Name              IP Address
XCAPI-SIP           172.16.0.153
procr               172.18.0.242
procr6              ::
anynode             172.18.0.100
ASM                 172.18.0.200
default             0.0.0.0
default-gateway     172.18.0.1

( 6 of 6   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
```

3.4 Codec Sets

This example uses **ip-codec-set 1** as shown next. It's recommended to use conform codec sets if running multiple network regions.

```
display ip-codec-set 1                                         Page   1 of   2

                          IP CODEC SET

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711A            n          2        20
 2: G.711MU           n          2        20
 3:
 4:
 5:
 6:
 7:
```

For additional configuration hints please check the chapters **Softfax (G.711 fax pass through)** starting on page 16 or **T.38** starting on page 17.

```
display ip-codec-set 1                                         Page   2 of   2

                          IP CODEC SET

                              Allow Direct-IP Multimedia? n

                                                                Packet
                          Mode                  Redundancy      Size(ms)
    FAX                   off                    0
    Modem                 off                    0
    TDD/TTY               off                    0
    H.323 Clear-channel   n                      0
    SIP 64K Data          n                      0              20
```

3.5 IP Network Region

The **ip-network-region** specifies the relations of the within- and between-region connectivity in the given IP region and its related VoIP resources and endpoints. The IP network region is here used as shown next.

```
display ip-network-region 99                                    Page   1 of   20
                               IP NETWORK REGION
  Region: 99
Location: 1       Authoritative Domain: te-systems.de
    Name: XCAPI-SIP              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      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

```
display ip-network-region 99                                    Page   3 of   20
                               IP NETWORK REGION

INTER-GATEWAY ALTERNATE ROUTING / DIAL PLAN TRANSPARENCY
 Incoming LDN Extension:
 Conversion To Full Public Number - Delete:    Insert:
 Maximum Number of Trunks to Use for IGAR:
 Dial Plan Transparency in Survivable Mode? n

BACKUP SERVERS(IN PRIORITY ORDER)    H.323 SECURITY PROFILES
 1                                   1   any-auth
 2                                   2
 3                                   3
 4                                   4
 6                                   Allow SIP URI Conversion? y

TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS
   Near End Establishes TCP Signaling Socket? y
                       Near End TCP Port Min: 61440
                       Near End TCP Port Max: 61444
```

```
display ip-network-region 99                                    Page   4 of   20

 Source Region: 99    Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set    WAN  Units     Total Norm  Prio Shr Regions          CAC  R  L    e
 1    1      y    NoLimit                                             n       t
 2
 3
```

3.6 Trunk Group

For this example the **trunk-group 99** is used as shown next.

```
display trunk-group 99                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 99                   Group Type: sip           CDR Reports: y
  Group Name: XCAPI-SIP                   COR: 1        TN: 1        TAC: #99
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                           Member Assignment Method: auto
                                                    Signaling Group: 99
                                                  Number of Members: 30
```

The **TRUNK PARAMETERS** on the second page are used as follows.

```
display trunk-group 99                                          Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                          Redirect On OPTIM Failure: 5000

            SCCAN? n                             Digital Loss Group: 18
                     Preferred Minimum Session Refresh Interval(sec): 90

 Disconnect Supervision - In? y  Out? y


             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n



 Caller ID for Service Link Call to H.323 1xC: station-extension
```

The third page of the trunk-group configuration dialog is used to modify some features, such as the **Numbering Format** parameter, which are here used as shown below.

```
display trunk-group 99                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                        Maintenance Tests? n


                     Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                                  Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no


 Show ANSWERED BY on Display? n
```

Via the **PROTOCOL VARIATIONS** settings, on the fourth page of the trunk-group configuration dialog, some protocol properties might be adjusted upon your needs. Here, the **Telephone Event Payload Type** is used with its default value **101**. The settings of the **IP DTMF TRANSMISSION MODE** parameter within the **system-parameters ip-options** should be also reviewed.

```
display trunk-group 99                                          Page   4 of  21
                                    PROTOCOL VARIATIONS

                                              Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                              Send Transferring Party Information? y
                                         Network Call Redirection? y

                                            Send Diversion Header? y
                                          Support Request History? n
                                     Telephone Event Payload Type: 101


                               Convert 180 to 183 for Early Media? y
                         Always Use re-INVITE for Display Updates? y
                               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: called-number-only
```

The **members** were added for the XCAPI trunk as determined initially.

```
display trunk-group 99                                          Page   5 of  21
                                         TRUNK GROUP
                                              Administered Members (min/max):   1/30
GROUP MEMBER ASSIGNMENTS                          Total Administered Members:  30

       Port             Name
  1: T00057             XCAPI-SIP
  2: T00058             XCAPI-SIP
  3: T00059             XCAPI-SIP
  4: T00125             XCAPI-SIP
  5: T00126             XCAPI-SIP
  6: T00127             XCAPI-SIP
  7: T00128             XCAPI-SIP
  8: T00129             XCAPI-SIP
  9: T00152             XCAPI-SIP
 10: T00153             XCAPI-SIP
 11: T00154             XCAPI-SIP
 12: T00155             XCAPI-SIP
 13: T00156             XCAPI-SIP
 14: T00157             XCAPI-SIP
 15: T00158             XCAPI-SIP
```

3.7 Signaling Group

The **signaling-group 99** is used as shown below. In accordance to the XCAPI configuration, the transport type is set to TCP. The **Near-** and **Far-end Listen Port** is used with the default port value 5060.

```
display signaling-group 99                                     Page   1 of   2
                                SIGNALING GROUP

 Group Number: 99             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: Others
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? n
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? y
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                   Far-end Node Name: XCAPI-SIP
 Near-end Listen Port: 5060                  Far-end Listen Port: 5060
                                          Far-end Network Region: 99

Far-end Domain: te-systems.de
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: allow                      RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? n
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y
                                                  Alternate Route Timer(sec): 6
```

3.8 Route Pattern

The **route-pattern** has to be related to the according trunk group.

```
display route-pattern 99                                       Page   1 of   3
                    Pattern Number: 99     Pattern Name: XCAPI-SIP
    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: 99   0                   1                                         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            unre                               unk-unk   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
```

3.9 AAR Analysis

The **AAR DIGIT ANALYSIS TABLE** is used for routing calls within your company's own private networks. For this example we use prefix **99** which is related to **route-pattern 99**.

```
display aar analysis 99                                         Page   1 of   2
                                  AAR DIGIT ANALYSIS TABLE
                                       Location: all             Percent Full: 0

              Dialed             Total     Route     Call     Node   ANI
              String            Min  Max  Pattern    Type     Num    Reqd
        99                      2    28    99        aar             n
                                                                     n
                                                                     n
                                                                     n
```

3.10 Feature Access Codes

This environment makes use of the feature access codes. So prefix 9 is used for accessing the XCAPI trunk. Along with the aar analysis any matching numbers starting with prefix **99** will be routed to XCAPI.

In reference to the numbering requirements there might be additional configuration tasks such as **uniform-dialplan, private-numbering, ars analysis, public-unknown-numbering** or others.

```
display feature-access-codes                                    Page   1 of  10
                                  FEATURE ACCESS CODE (FAC)
             Abbreviated Dialing List1 Access Code:
             Abbreviated Dialing List2 Access Code:
             Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                              Announcement Access Code:
                               Answer Back Access Code:
                                 Attendant Access Code:
         Auto Alternate Routing (AAR) Access Code: 9
       Auto Route Selection (ARS) - Access Code 1: 0     Access Code 2:
                        Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA:        All:        Deactivation:
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                                 Call Park Access Code:
                               Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
                          CDR Account Code Access Code:
                                Change COR Access Code:
                           Change Coverage Access Code:
                 Conditional Call Extend Activation:        Deactivation:
                      Contact Closure    Open Code:          Close Code:
```

3.11 Dial Plan Analysis

As required, the feature access code is referenced within the dial plan analysis.

```
display dialplan analysis                                         Page   1 of  12
                                DIAL PLAN ANALYSIS TABLE
                                     Location: all            Percent Full: 1

    Dialed    Total  Call       Dialed    Total  Call       Dialed    Total  Call
    String    Length Type       String    Length Type       String    Length Type
   0            1    fac
   1            3    ext
   2            3    ext
   3            3    ext
   4            1    fac
   5            1    fac
   6            1    fac
   9            1    fac
   *            3    dac
   #            3    dac
```

Configuration Notes

This chapter gives some configuration hints for appropriate interworking. Most of those XCAPI controller settings and configurations are set by default via the XCAPI controller wizard. Nevertheless they should be reviewed just as the according gateway parameters for appropriate interworking.

4.1 Clock Source

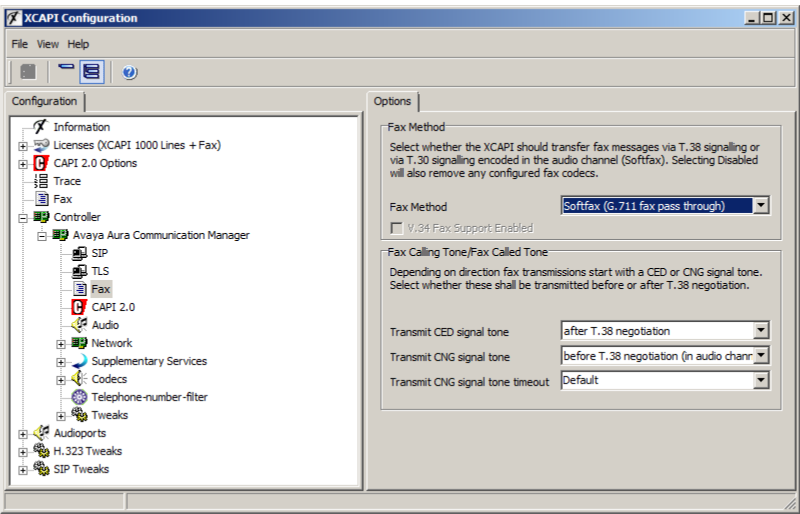
It's necessary that all layers are synchronized, especially for fax operations. Wrong synchronization evokes packetloss and leads to fax abruptions.

```
G450-001(super)# show sync timing
SYNCHRONIZATION CONTROL: --- Local ---

SOURCE     MM or VoIP           STATUS                    FAILURE
---------  -------------------  -----------------------  ---------------
Primary    v2                   Active                    None
Secondary                       Not Configured
Local      v0                   Standby                   None

Active Source: v2               Sync Source Switching: Enabled
```

4.2 Softfax (G.711 fax pass through)

With the Softfax mode, the XCAPI simulates an analogue fax device by transmitting modulated fax-signals modem-like through the established G.711 audio channels.
For this please check the XCAPI controller configuration tab labeled **Fax** and ensure that **Softfax (G.711 fax pass through)** is selected as **Fax Method**.

Ensure that the fax parameters of the ip-codec set(s) are configured as shown in chapter **Codec Sets** starting on page 9.

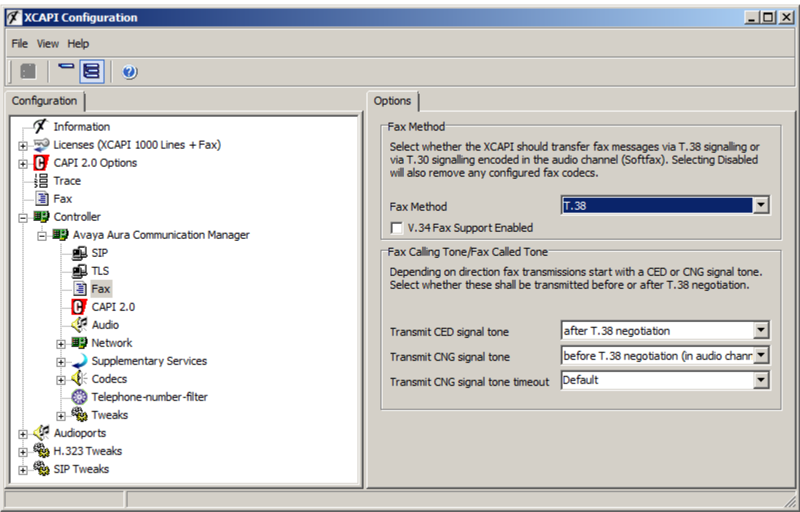


4.3 T.38

For enabling T.38 interworking the **IP Codec Set** has to include the G.711 codecs (at least one of them) and also the T.38 codec for appropriate codec negotiation. Ensure that the gateway DSP supports ECM (Error Correction Mode).
If ECM isn't available by the gateway DSP, Softfax (G.711 fax pass through) has to be preferred.

```
display ip-codec-set 1                                     Page   2 of   2

                                    IP CODEC SET

                                        Allow Direct-IP Multimedia? n

                                                                  Packet
                        Mode                Redundancy            Size(ms)
    FAX                 t.38-standard        2           ECM: y
    Modem               off                  0
    TDD/TTY             off                  0
    H.323 Clear-channel n                    0
    SIP 64K Data        n                    0                    20
```

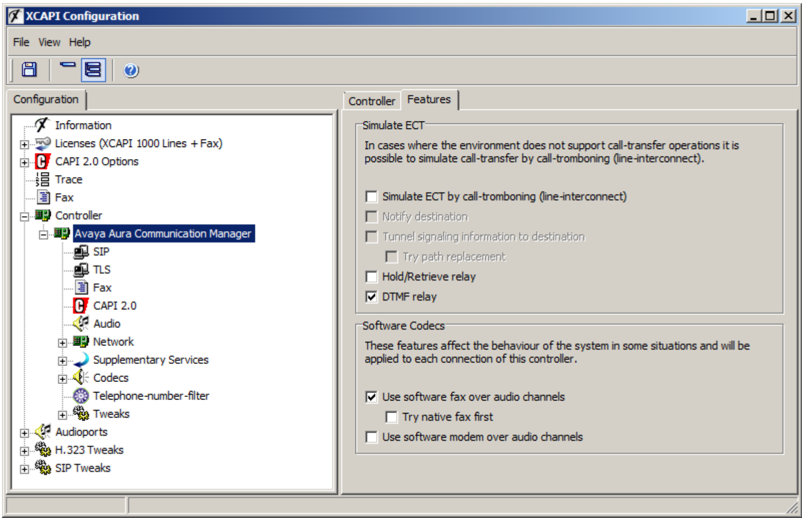
The XCAPI controller has to be set for the T.38 **Fax Method** as shown below.



For T.38 interworking, please ensure that both **Modem** and **TDD/TTY** are set to **off** and the involved IP codec sets are used conform. For T.38 or G.711 fax pass through troubleshooting, please check the XCAPI related SIP trunk with a **list trace tac** about the involved region and codec negotiation.

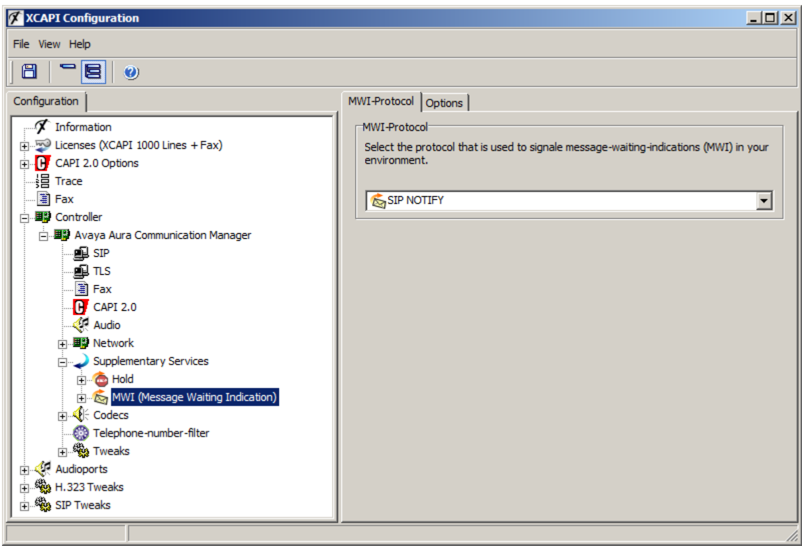
4.4 Call Transfer

For enabling call transfer via SIP refer please ensure that the **Simulate ECT by call-tromboning (line-interconnect)** is disabled within the XCAPI controllers **Features** tab. Ensure that the system parameters, class of restrictions and class of services are configured properly.



4.5 Message Waiting Indications

For MWI, please ensure that the **SIP NOTIFY** method is enabled within the XCAPI controller configuration for message waiting interoperability.



4.6 Redirection Number

Several CAPI applications need to receive a redirection number, in meaning of the gateway generated SIP diversion header, beside of the origins calling number. For this the **Send Diversion Header** must be enabled on page 4 of the XCAPI trunk group, see chapter **Trunk Group** starting on page 11.

Please note, XCAPI also support the **History-Info** header. On demand this can be evoked with enabling **Support Request History? y**.

```
display trunk-group 99                                        Page   4 of  21
                                PROTOCOL VARIATIONS

                                         Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? y
                                  Network Call Redirection? y

                                     Send Diversion Header? n
                                   Support Request History? y
                              Telephone Event Payload Type: 101


                        Convert 180 to 183 for Early Media? y
                  Always Use re-INVITE for Display Updates? y
                        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: called-number-only
```

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TE-SYSTEMS GmbH

Managing Directors Andreas Geiger
Oliver Körber

Address Max-von-Laue-Weg 19
D-38448 Wolfsburg
Germany

Tel. +49 5363 8195-0
Fax +49 5363 8195-999

E-Mail info@te-systems.de
Internet www.te-systems.de
www.xcapi.de