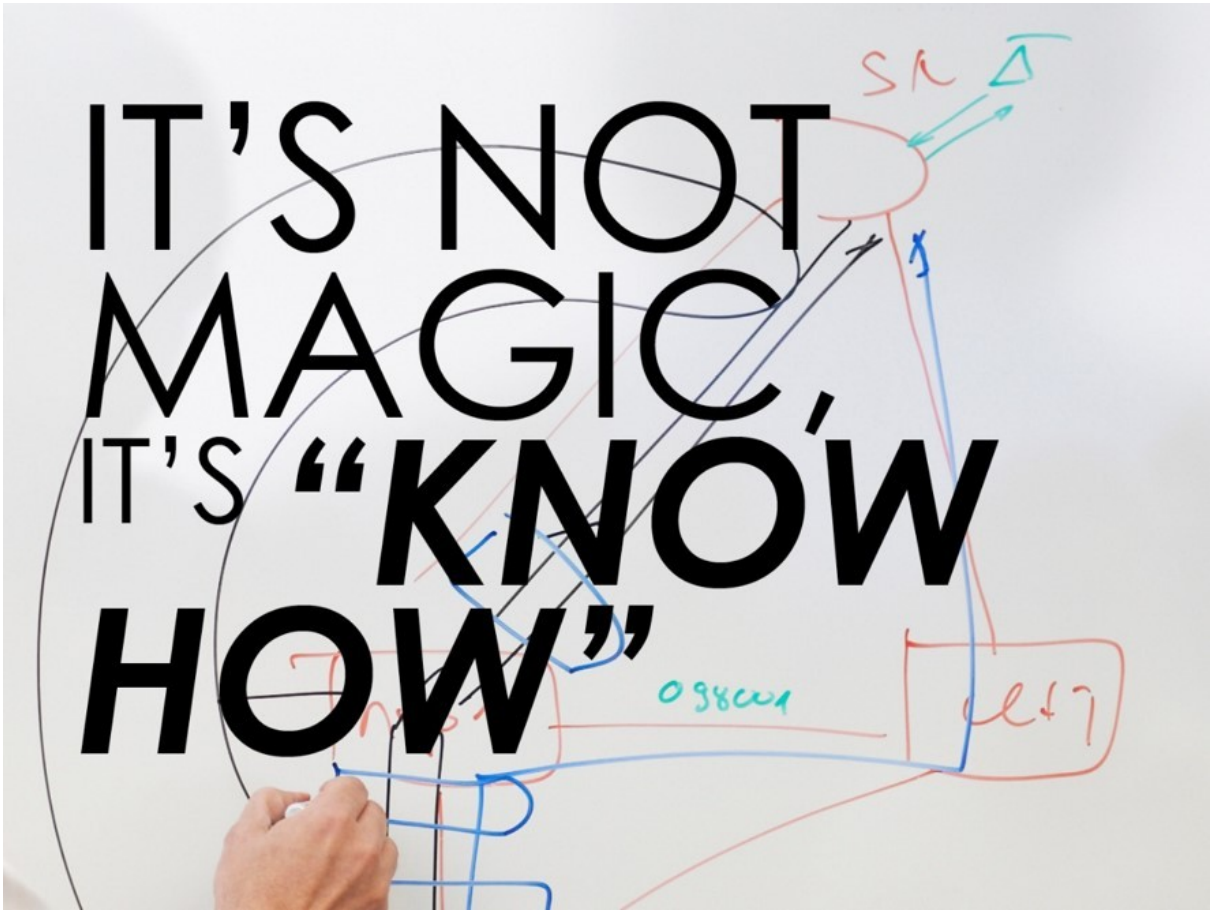
Book VoIP Switch Support



Introduction
The Aarenet VoIP System supporting personnel find here links to detailed information about:

◊ How to support telephony users and solving user problems
◊ An introduction to the VoIP signaling protocols
◊ The Aarenet VoIP Switch on board support tools
◊ The Aarenet VoIP System monitoring and alarming
◊ The maintenance and problem solving of the Aarenet VoIP Switch
◊ The maintenance and problem solving of DELL server

User Guide for Solving Telephony Problems (Support Level 1)

Best Practice: How do I start to solve my problem?

Best Practice

1. Check that the telephone (VoIP device) is connected correctly
   → Show me how...
2. You are trying to get the phone (VoIP device) up and running, but now:
   - ♦ It doesn't load its configuration from the VoIP Switch.
     → Show me how to check this ...
3. You have already been able to make (inbound and outbound) phone calls with the VoIP device, but now:
   - ♦ The telephone shows on its display that it has no connection to the VoIP Switch
   - ♦ The telephone (VoIP device) is no longer available for incoming calls
     → Show me how to check this ...
4. You have already been able to make (inbound and outbound) phone calls with the VoIP device, but now:
   - ♦ You can no longer establish or receive connections
     → Show me how to check this ...
5. You can make (inbound and outbound) phone calls with the VoIP device, but now:
   - ♦ The voice quality is very poor
   - ♦ My counterpart doesn't hear me
   - ♦ I can't hear my counterpart
6. We don't hear each other
   → Show me how to check this ...

Basic Check of the VoIP Device

The following basic conditions must always be checked first:

1. Is the VoIP terminal correctly connected to the power supply?
   Possible actions:
   - ♦ Replace the power supply cable
   - ♦ Use a different outlet
2. Is the VoIP terminal correctly connected to the data network?
   · Plug the data cable into the correct port:
   - ♦ At the VoIP device
   - ♦ At the upstream IP device (IP router, xDLS modem, etc.)

   Possible actions:
   · Replace the data cable
3. Does the VoIP device display the behavior and indicators described in its user manual?
   If it's not:
   Possible actions:
   - ♦ Contact the seller's or device manufacturer's support
4. Does your data network connection work?
   If your Internet connection for your PC and the VoIP device is running via the same upstream device (xDSL modem, FTTH modem (fibre optic modem), cable modem), can the Internet be accessed via your PC?
   I don't know or if no:
   Possible actions:
   - ♦ Contact the support of the Internet provider

Warning	• Defect power cables must be replaced! • Faulty power cables can be **life-threatening**!

The VoIP Device doesn't load the Configuration from the VoIP Switch

Note	These instructions are only valid if: 1. You have an account for the provider's self-care GUI. 2. The VoIP device type can be configured via the provider's self-care GUI. In case of doubt, contact the provider's support. For all other cases, the VoIP device must be configured according to the manufacturer's specifications!

How does the problem manifest itself:
You are trying to use the VoIP device for the first time or after a restart of the device with the default factory configuration, but:

- Nothing's working!

Verify in the user account of the self-care GUI:

→ Tab "Phones"
Click on the Button [State...]
◊ The "Last Access" parameter does not display the date/time and IP address of the VoIP device.
◊ The "MAC provisioning" parameter does not display "done".

Check the following conditions and take action:

1. Are the basic conditions fulfilled?
2. Is the VoIP device getting an IP address after connecting to the network?
   Possible actions:
   · Check network connection
   · Check DHCP service in your local IP network
3. Is the web based user interface of the VoIP device accessible and can you log in?
   Possible actions:
   · Check network connection
   · If the DHCP service is switched on in your local IP network, check via the telephone user or console interface whether the VoIP device obtains its IP address via DHCP.
4. Is the configuration of the VoIP terminal in factory defaults?
5. If no:
   Possible actions:
   · Restart the device manually with the default factory configuration (see the user manual of the VoIP device)
6. Is the VoIP Switch available?
   or is the redirection server of the device manufacturer configured correctly?
   or is the redirection server of the device manufacturer reachable?
   Possible actions:
   · Contact the support of the telephony provider

If the problem cannot be solved, contact the provider's support with the following information:

◊ Telephone number of the device which causes problems
◊ Device type
◊ Date and time when the problem occurred
◊ Description of the problem:
"The VoIP device cannot load its configuration!"

The VoIP Device doesn't Register with the VoIP Switch

Note	These instructions are only valid if: 1. You have an user account for the provider's self-care GUI. For all other cases, the VoIP device must be configured according to the manufacturer's operating instructions with the correct SIP credentials of your telephony provider !

How does the problem manifest itself:
The telephone (VoIP device) is used for the first time or it has already been possible to make a phone call (incoming and outgoing), but:

- Neither an incoming nor outgoing connection can be established with the VoIP device.
- An outgoing connection can be established with the VoIP terminal, but it is not possible to reach it inbound.

Verify in the user account of the self-care GUI:

→ Tab "Phones"
Click on the Button [State...]
◊ At "Registrations" no user agent, no IP address, no contact is displayed.

Check the following conditions and take action:

1. Are the basic conditions fulfilled?
2. Did the VoIP device load the configuration?
3. If no:
   Possible actions:
   · Configure the device manually or via the VoIP Switch.
4. What does the log of the VoIP device show?
   Possible actions

◊ Act according to the instructions of the VoIP device.

If the problem cannot be solved, contact the provider's support with the following information:

◊ Telephone number of the device which causes problems
◊ Device type
◊ Date and time when the problem occurred
◊ Description of the problem:
"The VoIP device cannot register!"

The VoIP Device cannot Establish or Receive Connections

How does the problem manifest itself:
The telephone (VoIP terminal) has already been able to make (incoming and outgoing) calls, but now:

- The VoIP device cannot establish or receive connections to/from public or private vPBX phone numbers whose devices are verifiably working (e. g. checked with an mobile phone).

Check the following conditions and take action:

1. Are the basic conditions fulfilled?
2. Has this VoIP device registered?

Possible actions:
- Check if the device is registered on the VoIP Switch .

3. For problems with incoming connections:
   - Is a call forwarding active?

   Possible actions:
   - Check with *#00 if a call forwarding is configured and deactivate with *00 if necessary.
   - If your VoIP device has a private vPBX phone number, have the vPBX administrator check if the call distribution is still working correctly.
4. For problems with outgoing connections:
   - Check with another device, e.g. a mobile phone or other phone of the same vPBX, if the desired destination number is reachable.
5. If no:

   Possible actions:
   - If you have a public phone number:
     - Check if a TopStop has been exceeded?
     - Is the desired destination number blocked by a RuleSet?
     - Contact the support of the telephony provider
   - If you have a private vPBX number:
     - Check if you need a leading 0, an other digit or no digit for outgoing calls to the PSTN.
     - Check if a TopStop has been exceeded?
     - Is the desired destination number blocked by a RuleSet?
     - Contact the vPBX administrator

If the problem cannot be solved, contact the provider's support with the following information:

◊ Telephone number of the device which causes problems
◊ Date and time when the problem occurred
◊ Telephone numbers of the participating devices:
A Number of the calling side
B Number of the called side
◊ Description of the problem:
"A cannot make calls"
"A cannot receive calls"
"A cannot make calls to certain B numbers:"
◊ List of B numbers that cannot be called

Poor, Partly or Completely Missing Speech Transmission

How does the problem manifest itself:
The VoIP device can establish or receive connections. The voice transmission was fine on earlier connections, but not now.

- The voice transmission is disturbed:

  ◊ B hears A disturbed
  ◊ A hears B disturbed
  ◊ The speech transmission is disturbed in both directions A <-> B

- The voice transmission is missing in part or in full since beginning of the connection:

  ◊ B does not hear A
  ◊ A does not hear B
  ◊ A and B do not hear each other

Check the following conditions and take action:

1. Are the basic conditions fulfilled?
2. Is the handset or headset connected correctly?
3. Is the microphone of the handset or headset not muted?
4. Are the volume levels for the loudspeaker and microphone on your telephone set correctly?
5. Is the problem only with a specific parter? If yes, the other party should check the volume of the microphones, headset and headset being used.
6. Remember:
   - Handsfree mode often produces poor voice quality.
   - Connections with mobile phones can be disrupted, especially when the call partner is traveling.

If the problem cannot be solved, contact the provider's support with the following information:

◊ Telephone number of the device which causes problems
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◊ Description of the problem:
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FAX Transmissions do not or Only Partially Work

In a VoIP environment FAX no longer achieve the same degree of reliability as before in an analogue or ISDN one. The FAX reliability depends on various factors such as the type of device, device settings and the way the device is connected to the IP network. It depends also on the quality of the transmitter and receiver of the peer FAX devices. Getting all these factors together a transmission may not even start or dropped unexpectedly. The transmitted documents may be incomplete.

The users must expect increasing difficulties in the future, especially for international transmissions.

How does the problem manifest itself:

- FAX transmission doesn't start
- FAX transmission is dropped
- The transmitted document is incomplete

Check the following conditions and take action:

1. Adjust the FAX device configuration:
   - ♦ Reduce the transmission speed to max. 9600bds.
   - ♦ Switch OFF the error correction, e.g. EMC
   - ♦ If the device offers a "VoIP mode" then experiment with it and check if the results are better.

If the problem cannot be solved, contact the provider's support with the following information:

◊ Telephone number of the device which causes problems
◊ Device type
◊ Date and time when the problem occurred
◊ Telephone numbers of the participating devices:
A Number of the calling side
B Number of the called side
◊ Description of the problem:
- "FAX Transmissions doesn't work"

Don't expect miracles from the support!

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5. If no:
   Possible actions:
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   ♦ Check if a TopStop has been exceeded?
   ♦ Is the desired destination number blocked by a RuleSet?
   ♦ Contact the support of the telephony provider
   · If you have a private vPBX number:
   ♦ Check if you need a leading 0, an other digit or no digit for outgoing calls to the PSTN.
   ♦ Check if a TopStop has been exceeded?
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   ♦ Contact the vPBX administrator

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6. Remember:
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◊ Telephone numbers of the participating devices:
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◊ Description of the problem:
- "FAX Transmissions doesn't work"

Don't expect miracles from the support!

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Introduction for Supporting User Problems (Support Level 2)

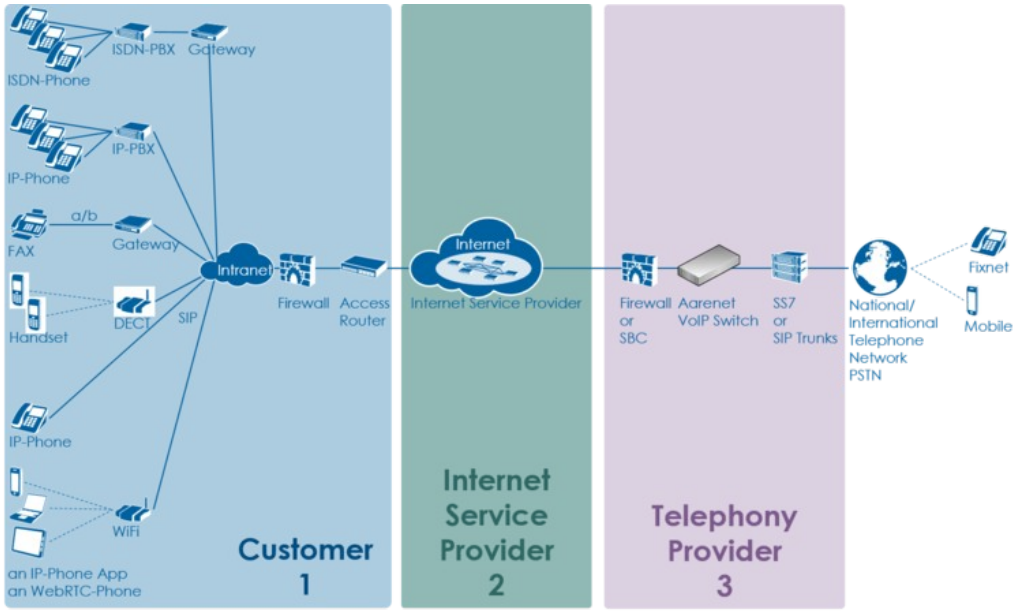
Introduction Support Level 2

The level 2 support is the first instance where the user's telephony problems are handled that a user cannot solve himself. Additionally the level 2 supporter must be able to detect if the user problem is a "single" problem or if there is a large scale problem, that produces the same problem for multiple customers, e.g. data transfer problems in the Internet so that no VoIP call signaling is possible.

The level 2 supporter must be aware of the complexity of a VoIP system and the multitude of telephony solutions on the user side. Further he needs an understanding of:

◊ IP networking
◊ VoIP protocols SIP for signaling, SDP and RTP for speech transmission.

Overview of a VoIP system and the multitude of user telephony solutions:



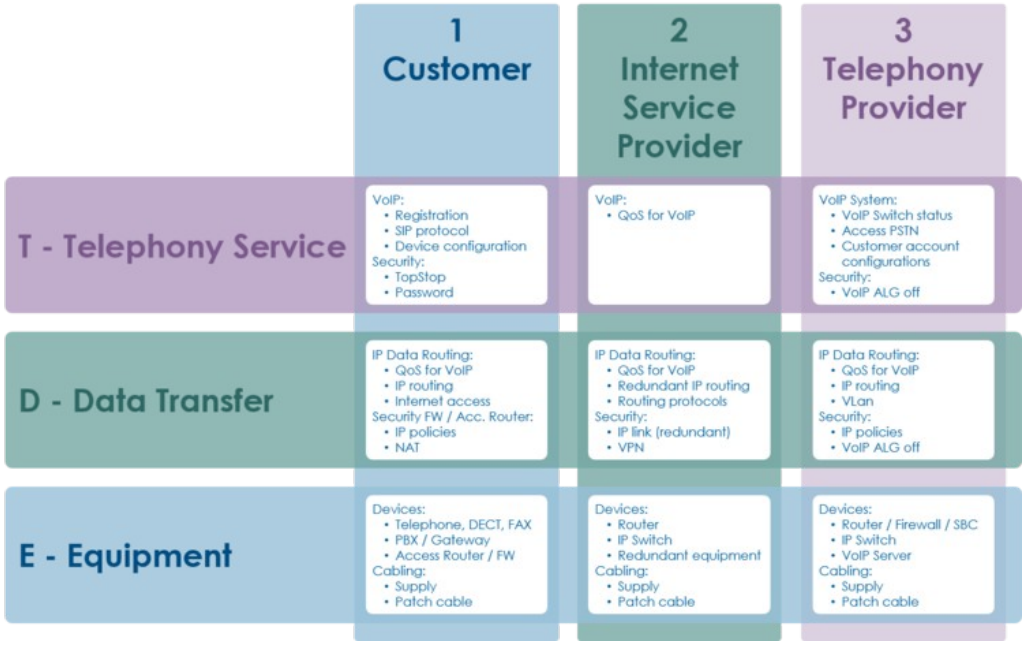
The level 2 supporter faces problems with the following layers:

1. Equipment
2. IP data transfer
3. Telephony service

And each of this layer can be located into the following raw areas:

1. Customer/User site
2. Internet Service Provider ISP
3. Telephony Provider

This layer and dividing into areas produces a "3x3 Support Matrix":



Within this "3x3 Support Matrix" the supporter can:

◊ advice the customer what to do when the problem is located in the nodes 1-T, 1-D and 1-E
◊ check the customer configurations on the VoIP Switch, node 3-T, and, if he has operator rights, adjust configurations.

For the other cases the level 2 supporter must be able to identify if:

◊ the user must contact his ISP, due to possible Internet access problems
◊ the VoIP System administrator must be involved, due to possible telephony service problems

Hint:
These cases indicate mostly large scale problems within the VoIP system!

Best Practice for Handling an User Problem

Best Practice

1. Record the customer's data and problem description:
   - ♦ Get the customer data
   - ♦ Get the problem description from the customer
   - → Show me how ...

2. Cross check the user input against the VoIP switch configuration and logs.
   Via ConfigCenter check the users account configuration:
   - · Telephone number registration
   - · TopStop
   - · RuleSet
   - · Consult the "Call Data" for the last connection attempts and connections longer than 2min

   → Show me how ...

3. Evaluate the user's VoIP setup:
   - ♦ For questioning and analyzing the user's problem it is necessary that the supporter is aware of the VoIP setup of the user.
   - → Show me how ...

4. Check the big picture:

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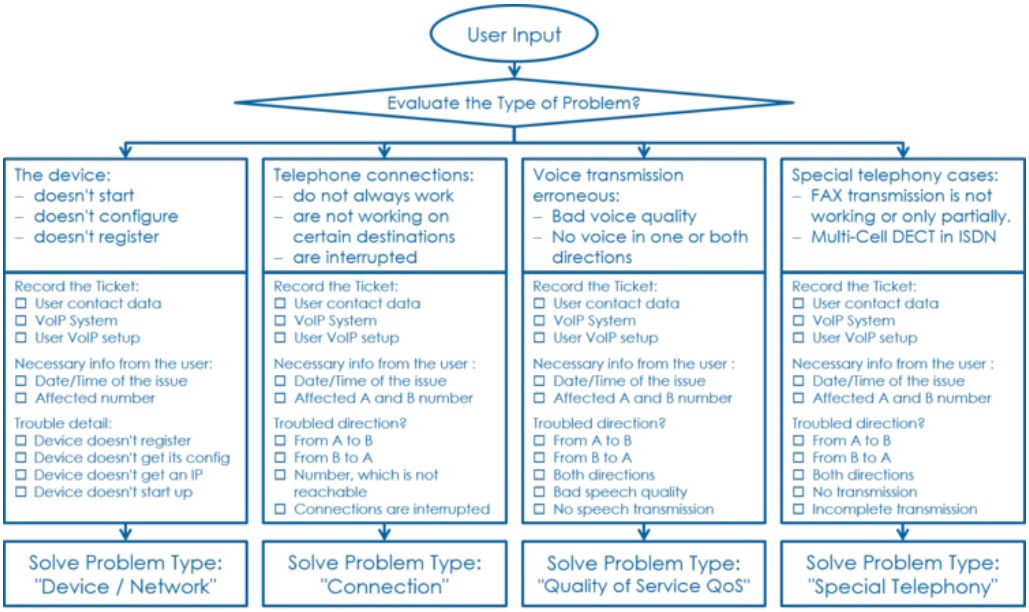
- If you are sure that the problem is "single" one continue with the next step "Solve the customer problem"
- If you suspect that the customer is not the only one with this problem within a short time range, e.g. 30min, then contact the VoIP switch administrator/operator if there is known issue in the VoIP system that causes this type of problem.
  → Show me how ...

5. Solve the customer's problem:
   - You have now enough basic information for solving the customer's problem
     → Show me how ...

6. If you are not able to solve this problem then contact the provider support. Have ready all collected information and what you have done and found out until now.

The supporter shall record the user information and the results of the own research:



Note	This information is most welcome if the supporter needs the support of the provider and has to inform him about the case.

Step 1: Record the Customer's Data and Problem Description

Get the Customer's Data

From the customer get:

◊ Name of the caller
◊ Telephone number of the caller
◊ if applicable the company name

Write down the Customer's Problem Description

From the customer get:

◊ Date and time of the issue
◊ The involved telephone numbers
◊ The problem description

If the customer doesn't know then identify via the ConfigCenter the telephone number and its associated account.

Step 2: Cross Check the User Inputs

With this cross check the supporter can validate the user information, gets an impression of the state of the account and will probably find the reason for the user problem...

Via ConfigCenter check the users status and account configuration on the VoIP Switch:

1. Check "Validity":
   Check if the user account and its addresses are existing and "valid".
2. Check the telephone number registration status.
   If there is no registration you can proceed directly with → The user device is not registered ...
3. Check "TopStop":
   Check if a TopStop in the account or address prevents the user from doing outgoing calls.
4. Check "RuleSet":
   Check if a selected RuleSet in the account or address configuration prevents the user from doing outgoing or receiving calls.
5. Check "Call Forwards" or "Call Rejecting":
   Check if a "Call Forwards" or "Call Rejecting" in the account prevents the user from doing outgoing or receiving calls.
6. Check "Call Data":
   Consult the "Call Data" for the last connection attempts and connections longer than 2min of the user.

Step 3: Evaluate the User's VoIP Setup

For questioning and analyzing the user's problem it is necessary that the supporter is aware of the VoIP setup of the user.

The experienced supporter knows of the user's VoIP setup after the cross check . If not here the supporter finds the most implemented VoIP setup's.

VoIP Setup: Residential

Characteristics:

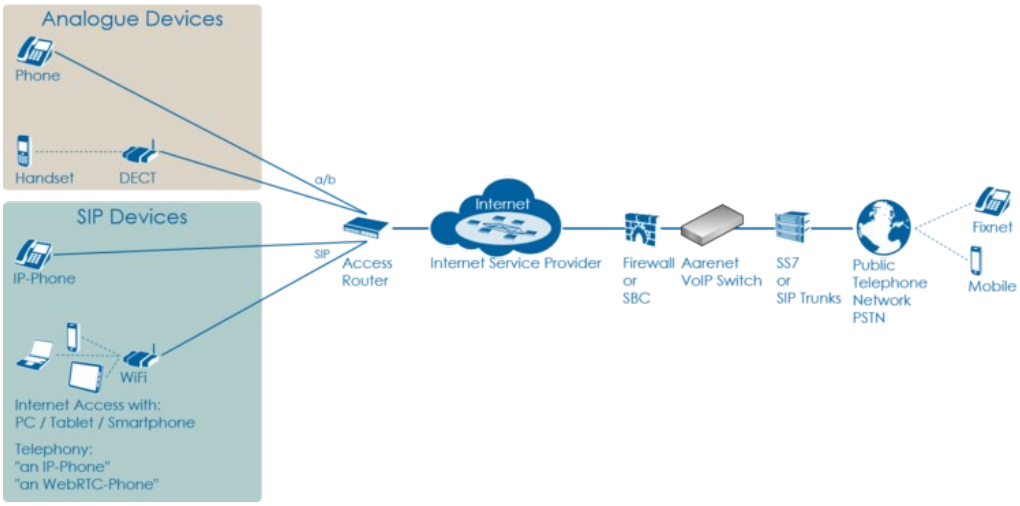
◊ Private household
◊ Single or few telephone number
◊ Each telephone number registers individually

Most common problems:

◊ Account or telephone number blocked on the VoIP switch
◊ Telephone number not correctly ported to the telephony provider
◊ Telephone not correctly configured

◊ Telephone, cables defect
◊ Internet access fails

Overview VoIP Setup:



VoIP Setup: Legacy ISDN PBX

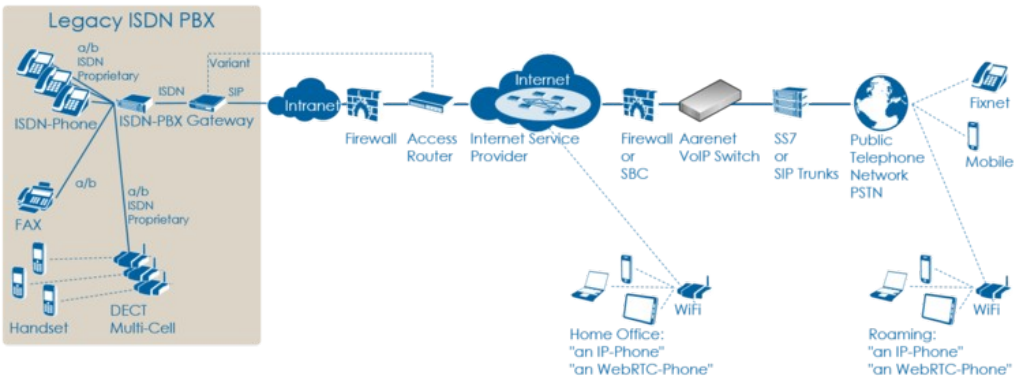
Characteristics:

◊ Company PBX
◊ The ISDN PBX is connected via BRI or PRI to an ISDN-SIP Gateway
◊ One or more telephone number ranges
◊ The telephone numbers are registered via a main number
◊ The telephone number of incoming calls are signaled with only a few digits

Most common problems:

◊ Account or telephone numbers blocked on the VoIP switch
◊ Telephone number ranges not correctly ported to the telephony provider
◊ Telephone number ranges not completely configured on the VoIP Switch
◊ Wrong incoming telephone number signaling
◊ Internet access fails
◊ The company Firewall VoIP ALG interferes with the SIP signaling or needed IP ports are blocked.
◊ QoS problems for speech, Fax, DECT

Overview VoIP Setup:



VoIP Setup: IP PBX

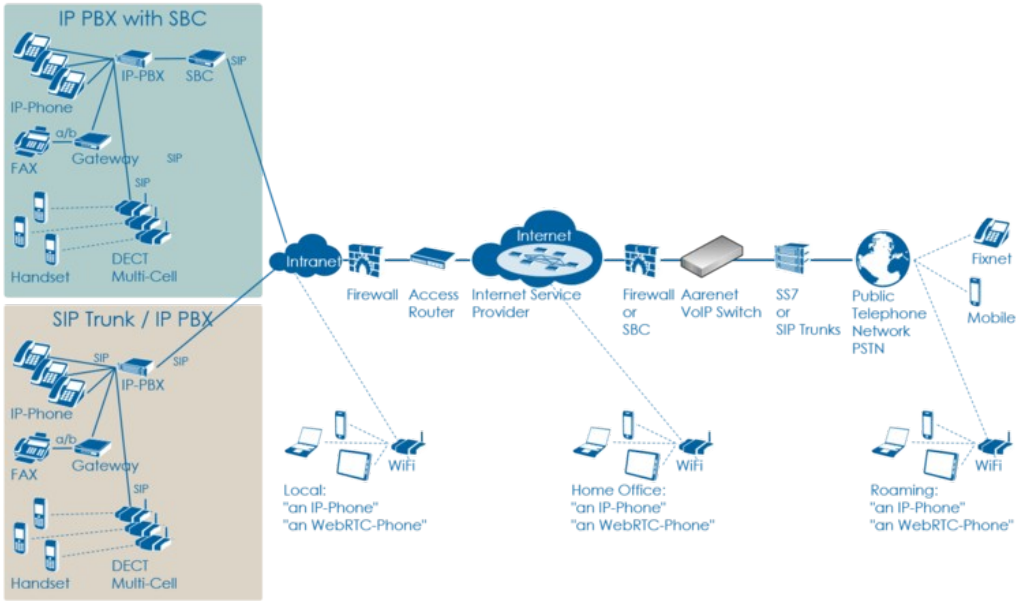
Characteristics:

◊ Company PBX
◊ The IP PBX is connected directly or via SBC to the VoIP Switch
◊ One or more telephone number ranges
◊ The telephone numbers are registered via a main number

Most common problems:

◊ Account or telephone numbers blocked on the VoIP switch
◊ Telephone number ranges not correctly ported to the telephony provider
◊ Telephone number ranges not completely configured on the VoIP Switch
◊ The company Firewall and/or SBC VoIP ALG interferes with the SIP signaling or needed IP ports are blocked.
◊ Internet access fails
◊ QoS problems for speech

Overview VoIP Setup:



VoIP Setup: vPBX

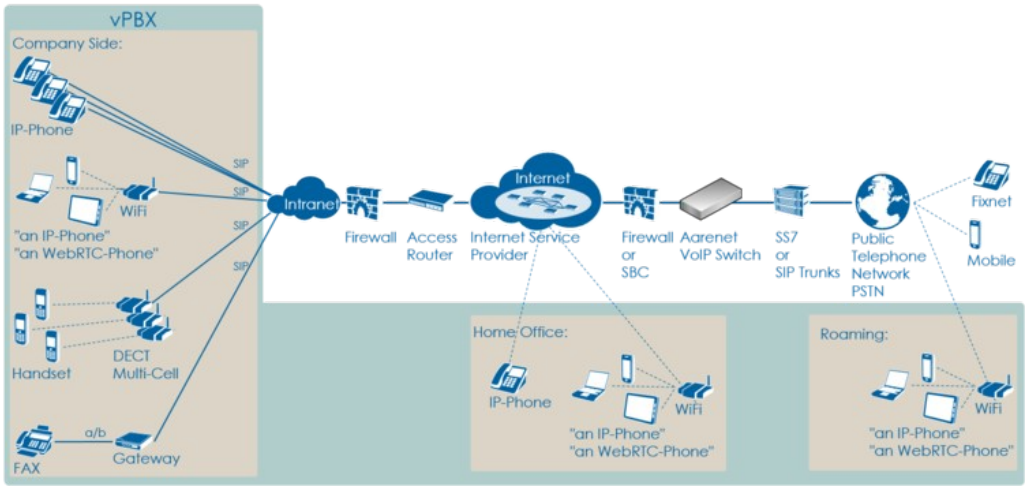
Characteristics:

◊ Company PBX
◊ The IP Phones are connected directly to the VoIP Switch
◊ One or more telephone number ranges

Most common problems:

◊ Public account and/or public telephone numbers blocked on the VoIP switch
◊ Public telephone number ranges not correctly ported to the telephony provider
◊ Telephone number ranges not completely configured on the VoIP Switch
◊ Private account and/or private telephone numbers blocked on the VoIP switch
◊ Provisioning of the SIP devices out of the AdminCenter
◊ The company or home office Firewalls and/or SBCs VoIP policies or ALG interferes with the SIP signaling or needed IP ports are blocked.
◊ Company/home office Internet access fails
◊ QoS problems for speech, FAX, DECT

Overview VoIP Setup:



Step 4: Check the "Big Picture"

At this point the supporter should get aware if the problem is limited to this user or if it could be large scale problem within the VoIP System.

If the supporter suspects a large scale problem, due to a great amount of the same ore similar user complains then he should contact the telephony provider support or emergency organization.

If the supporter has enough privileges he can check:

1. The VoIP Switch component status
   This will show if the VoIP Switch itself has a problem.
2. The VoIP System monitor
   Here you can check if:
   - The registrations dropped in a large scale
   - The calls dropped in a large scale
   - The IP connectivity somewhere in the VoIP system failed

At any rate the supporter **must inform** the VoIP system administrator!

Step 5: Solve the Customer Problem

Solve "Device / Network / Configuration / Registration" Problems

This problem type covers the following erroneous conditions:

- ◊ The device doesn't start
- ◊ The device doesn't integrate into the IP network
- ◊ The device is not correctly configured
- ◊ The device doesn't register at the VoIP Switch

Note	If the device is connected to an IP-PBX then these problems must be solved with the responsible of the IP-PBX.

Solve "Device Hardware & Firmware" Problem

1 Step: Is the device powered on, not defect?

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			
Equipment	Check if the device correctly powered and shows basic activity? ◊ Is the power cable correctly plugged in? ◊ Is the power cable not defect? ◊ Does the device show power on indication, e.g. display on, LED on? Actions: Replace the power cable Replace defect device if the powering is ok but no working indication is displayed		

Warning	Defect power cables must be replaced! Faulty power cables can be life-threatening!

2 Step: Is the device connected to the IP network?

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			
Equipment	Is the device correctly connected to the IP network? ◊ Is the patch cable correctly plugged in? ◊ Is the patch cable not defect? ◊ Are there LED flashing or glow next to the network plug on the device or at the peer device (access router, IP switch)? Actions: Replace the patch cable Plug in the patch cable at a different port at the peer device (access router, IP switch) Replace defect device if the patch cable and peer port is ok but no working indication is displayed at the device port.		

3 Step: Has the device a reasonable firmware loaded?

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			

Equipment	Has the device a reasonable firmware loaded? ◊ The user must check the loaded firmware Actions: Replace the firmware if outdated or important bugs are fixed		

Solve "Device Network" Problem

1 Step: Has the device an IP address and can access the Internet?

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer	Has the device got an IP address? ◊ Check on the device if it has received an IP address, e.g. via display or maintenance GUI. Actions: ◊ If no IP address was assigned the user must: ◊ Check if the device is really connected to the IP network! ◊ Check if the device is configured with a fixed IP address! If it has a fixed IP does it match with the IP subnet? Is the default GW and DNS entry configured? ◊ Check if the device is configured to use DHCP! if the DHCP service in the IP network is running If the user cannot check the DHCP service he must contact the company IT responsible or the responsible of maintaining the access router. Has the device access toward the VoIp Switch? ◊ Check if the device makes contact with the VoIP Switch via Internet or any private IP Link. Actions: ◊ If the device shall connect via the Internet: Connect a PC to the Ethernet port where the device usually is connected and try to connect to any public Web site. ◊ If the device shall connect via an private network check with the IT responsible if the access to the VoIP Switch is guaranteed.		
Equipment			

Solve "Registration" Problem

1 Step: Review the account and telephone number configuration

Customer	Internet ISP	Telephony Provider

Telephony			Check via ConfigCenter: ◊ Does the user account exist and is it "valid"? ◊ Does the telephone number exist and is it "valid"? Actions: ◊ Check why the account, telephone number doesn't exist or is disabled activate them if allowed.
Data Transfer			
Equipment			

2 Step: Where REGISTER messages received from the device on the VoIP Switch?

	Customer	Internet ISP	Telephony Provider
Telephony			In the "Support Log" search for the device registration in the present and past time. 1. Set the "Support Log" filters · Insert at "Text" the account name · Insert at "From" - "Until" a reasonable time span where registrations could be expected · Select the category: "Registration" 2. Start the search and find out according the results what is going wrong. 3. If needed repeat the search with the telephone number in the "Text" or other time spans ◊ Check the log results → see below Actions: ◊ → see below
Data Transfer			
Equipment			

Failed registrations due to disabled account or address:

	2017-09-15-07:56:49.553 Registration failed, disabled account aan1-00093 tried to register number 0449980010

Actions:

◊ Check why the account is disabled and activate if allowed.

Failed registrations due to wrong SIP credentials:

	2017-09-15-08:05:38.117 Registration failed, invalid credentials for account acc-01 2017-09-15-08:05:39.112 Registration failed, unknown username 'myusername' tried to register '0123456789' 2017-09-15-08:05:38.377 Registration failed, unknown number '0987654321' tried to register for account acc-01

Actions:

◊ The user must manually adjust the SIP credentials on the device
◊ The user must re-configure the device via AdminCenter

The device didn't refresh its registration:

	2017-09-15-07:59:00.862 RegID989961 ended for 0987654321 ip=111.111.111.111:65398 ua=my-device v1.0

Actions:

◊ Order the user to check if the device is really on-line!
◊ Order the user to check if the device is defect? powered on? patch? IP address? see below

For information a successful registration:

	2017-09-15-07:59:30.383 RegID989965 started for 0987654321 ip=111.111.111.111:65398 ua=my-device v1.0

Hint:
The supporter might try to find REGISTER messages from the device in the "Trace" . This gives the certainty that the message was received by the VoIP switch. The supporter can filter for the telephone number. If the IP address is needed then the customer must be able to tell or evaluate it, e.g.:

```
https://www.whatismyip.com/
```

3 Step: Is the device correctly configured for registration??

	Customer	Internet ISP	Telephony Provider
Telephony	For a manually configured device, check that the device has the correct configuration for: ◊ Telephone number ◊ SIP credentials ◊ VoIP Switch domain configuration Actions: The user must manually check the device configuration and if needed adjust its configuration of the telephone number, SIP credentials and VoIP Switch domain for registration For a automatically via AdminCenter configured device check that: ◊ the selected device type in the AdminCenter is identical to the physical one. Actions: If not the same type then the user must re-configure the device via AdminCenter For a automatically via AdminCenter configured device check that: ◊ the user device has downloaded its configuration. Actions: If the configuration is not downloaded then it must be checked if the device: ◊ has got an IP address in the local IP network ◊ has access to the Internet ◊ has access to the configuration download of the Telephony Provider. By default this is the IP address of the VoIP Switch domain and uses the protocol HTTPS on TCP port 443. Check with the Telephone provider or via ConfigCenter > Menu "System" > "Zone Profiles" The user must check if the Firewall, SBC or Access Router doesn't block HTTPS traffic to and from the configuration download of the Telephony Provider		

Data Transfer			
Equipment			

Solve "Connection" Problems

This problem type covers the following erroneous conditions:

◊ Incoming or outgoing calls are not working
◊ Wrong called number
◊ Call supervision
◊ User device not registered
◊ User device not correct configured
◊ SIP signaling in general

Note	If the device is connected to an IP-PBX then these problems must be solved with the responsible of the IP-PBX.

1 Step: Review the account and telephone number configuration / registration?

	Customer	Internet ISP	Telephony Provider
Telephony			Do this check for the A and/or B telephone number if they are on-net numbers of the VoIP SWitch. Check via ConfigCenter: ◊ Does the telephone number exist? ◊ Is the telephone number valid? ◊ Is the user account valid? ◊ Is the telephone number correctly registered Actions: Check why the account, telephone number doesn't exist or is disabled and activate if allowed. Check why the device is not registered at the VoIP Switch
Data Transfer			
Equipment			

Hint:

◊ If the device is not registered outgoing calls might be working but NO incoming call will work.

2 Step: Was the called number correctly transmitted to the peer?

	Customer	Internet ISP	Telephony Provider
Telephony			Check via ConfigCenter: 1. In the "Call Data" search for the erroneous call: ♦ Set the "Call Data" filters: · Insert at "Time" a reasonable time span where the erroneous call is to be expected.

			· Set "Duration" to 00:00:00 · Insert at "Called Number" the called number 2. Start the search and identify the CDR of the erroneous call in the list If no CDR was found search for the "Calling Number"! 3. Open the identified CDR and get the trace of the call, click the Button [Trace] 4. Check if the called number in the "TO-Header" in all INVITE messages is correct: ♦ Is the called number correct? Often the users don't dial all digits or wrong digits or the configured number on a direct call key is incorrect. ♦ If the number is dialed correctly then it can be that the destination is not reachable. ♦ Outgoing calls from a vPBX can miss the public prefix 5. Check the peers call cancel reason: ♦ SIP Failure Responses Actions: Inform the user to dial the correct number. Try to reach the called number via an alternative telephone network, e.g. from a mobile telephone. Check with the support of the telephony provider why the called number is not reachable.
Data Transfer			
Equipment			

3 Step: What is the reason of an interrupted connection?

	Customer	Internet ISP	Telephony Provider
Telephony	Search in the "Call Data" for the erroneous call: 1. ♦ Set the "Call Data" filters: · Insert at "Time" a reasonable time span where the erroneous call is to be expected. · Set "Duration" to 00:00:00 · Insert at "Called Number" the called number 2. Start the search and identify the CDR of the erroneous call in the list If no CDR was found search for the "Calling Number"! Open the identified CDR and check for the release or reject reason: ◊ Was the connection released by a peer, A or B side? ◊ Check cancel reason in "State" (SIP Failure Responses) Actions: Inform the user about the release reason, e.g. his own device or the peer device released the call regularly (but probably nor expected). Check if a call was released due to call supervision: ◊ Variant 1: Session Timer was not refreshed: · Open the "Trace" of the connection and check if the connection was released due to missing RE-INVITE from the peers when the Session Timer run out. Actions: Inform the user that his device did not restart the Session Timer. The device configuration must be inspected an adjusted if needed. ◊ Variant 2: SIP INFO were not answered by the peer:		

	· Open the "Trace" of the connection and check if the connection was released due to not answered INFO messages that were sent from the VoIP Switch toward the peers. If activated the INFO's are sent usually every 120sec. Actions: Inform the user that his device did not answered INFO messages. It must be checke with the support of the device manufacturer if the device doesn't send 200 ACK when an empty INFO message was received ("SIP ping"). ◊ Variant 3: Missing RTP packets between the peers: This type of problem is a difficult one and hard to check and solve! It must be handled like a QoS problem. Media transferring devices as the MediaServer of the VoIP Switch, SBCs, SS7-Gateways, SIP-Trunks supervise the media stream of RTP packets. If after a certain time no RTP packets are transferred in a connection then such an instance can release the call. Typically after 30secs a connection is released if no RTP streams are detected. ◊ · Open the "Media Trace" of the connection and check if there are remarkable differences between the amount of sent and received or lost RTP packets. Actions: → See "Quality of Service QoS problem" below.		
Data Transfer			
Equipment			

Solve "Quality of Service QoS" QoS-Problems

Introduction to QoS-Problems

Note	In most cases, QoS-problems can only be found and solved by means of an exclusion procedure. It is paramount that the customer/user knows that QoS-problems are difficult to track down and to solve. It's nerve-wracking and it is time consuming. Solving QoS-problems often requires the cooperation and active co-testing from the customer/user with the support personnel! The active help of the customer/user is needed in most cases, e.g. by executing test connections.

The QoS-problem type covers the following erroneous conditions:

◊ No voice transmission in one or both directions from the beginning of the connection
◊ Bad voice quality during the connection

Naming and characteristics of QoS-problem:

One/No-Way Connection:

There is no speech transmission in one or both directions from beginning of the connection:

- Silence (Possible reason: Mostly due to no or blocked RTP data transmission)

Glitch Connection:

There is speech transmission but it is disturbed:

- Crackle, clicking (Possible reason: small packet loss, jitter)
- Short interruption (Possible reason: bigger packet loss)
- Ouw-ing (Possible reason: jitter, transcoding)
- Echo (Possible reason: jitter, big delay)

The source of the QoS-problems are all too often somewhere in the data transmission "D Data Transfer" layer (but sometimes they are surprisingly simple):

◊ The microphone or loadspeaker in the telephone handset defect
◊ Volume configuration in the telephone set wrong
◊ Telephone defect
◊ The company Intranet is not made ready for VoIP
◊ Any device in the "D - Data Transfer" layer

1st Step: Interview the User

1 Step: Interview the user carefully and identify the type of QoS-problem

Get all information from the user:

1. Occurs the the QoS-problem with all peers or just with the given B peer?
   Hint:
   If the problem occurs only with the B peer then this is a strong indication that something is wrong on the B side!
2. Is there no voice transmission, neither from `A->B` nor `B->A`?
   Type of QoS-problem: "No-Way Connection"
3. Is there voice transmission from `A->B` (B hears you) but none from `B->A` (you don't hear B)?
   Type of QoS-problem: "One-Way Connection `A->B`"
4. Is there voice transmission from `B->A` (you hear B) but none from `A->B` (B doesn't hear you)?
   Type of QoS-problem: "One-Way Connection `B->A`"
5. Are there during the connection crackle, clicking, short interruptions, uow-ing in the voice transmission for both sides?
   Type of QoS-problem: "Glitch Connection"
6. Are there during the connection crackle, clicking, short interruptions, uow-ing in the voice transmission from `A->B`?
   Type of QoS-problem: "Glitch Connection `A->B`"
7. Are there during the connection crackle, clicking, short interruptions, uow-ing in the voice transmission from `B->A`?
   Type of QoS-problem: "Glitch Connection `B->A`"
8. Uses the user an ISDN or DECT telephone behind an ISDN-PBX? Does the user have sharp clicking glitches in a regular or irregular interval? Do experience all users behind this ISDN-PBX this clicking?
   Remember :
   This points to a synchronization problem of the ISDN-PBX!
9. Is one peer A or B a mobile user?
   Remember:
   Mobile networks often have QoS-problems on the wireless links between the base station and the mobile device!

Action:

Cross check the users information by checking the media transfer statistics of the affected connection, see "2 Step" below!

2nd Step: Localize the QoS-Problem

Note	It is very important that the supporter is aware of of the localization of the problem. QoS-problems in the range of the "2 Internet Service

	Provider ISP" or "3 Telephony Provider" will affect usually a lot of users immediately.

1 Step: Check the "Big Picture"

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer	Check with the ISP where the user is connected to if there are outages in: ◊ the ISP IP network Actions: Inform immediately the Telephone providers support or alarming organization. If yes, just wait until the outage is solved	Check with the ISP where the VoIP System is connected to if there are outages in: ◊ the ISP IP network ◊ relevant national and/or international IP network Actions: Inform immediately the Telephone providers support or alarming organization. If yes, just wait until the outage is solved	Check with the IT responsible of the IP network where the VoIP System is attached to: ◊ Are there known outages in the IP network where the VoIP System is attached to? ◊ Is there a large scale QoS-problem? ◊ Are users affected which are located: • in a certain private IP network of the telephony provider? • at a definable tenant? Actions: Inform immediately the Telephone providers support or alarming organization. If the VoIP System is located in a pure private IP network then contact immediately the IT responsible or IT emergency organization. If yes, just wait until the outage is solved
Equipment			

2 Step: Identify the disturbed transmission direction from the VoIP Switch's view

This identification bases upon the VoIP System setting that all media streams are routed via the MediaServer of the VoIP Switch. The MediaServer collects statistic information about all media stream that are routed through it. These statistics can help to identify the source of the QoS-problem.

Search in the "Call Data" the CDR of the erroneous call:

1. ♦ Set the "Call Data" filters:
   · Insert at "Time" a reasonable time span where the erroneous call is to be expected.
   · Set "Duration" to 00:00:00
   · Insert at "Called Number" the called number
2. Start the search and identify the CDR of the erroneous call in the list
   If no CDR was found search for the "Calling Number"!
3. Open the identified CDR
4. Get the RTP statistics of this connection, click Button [Media Trace]
   If there are no data in the "Media Trace" contained then the media stream is not routed via the MediaServer of the VoIP Switch. See below how to force the routing via the MediaServer.
   Depending of the identified QoS-problem type analyze the RTP statistics detail, see below

If the media stream are not routed by default via MediaServer the supporter can force it for an account via the ConfigCenter:

→ Menu "Account"

→ Select the customers account

→ Tab "Advanced"

→ Set "Use always MediaServer" to `"Yes"`

Note	◊ By forcing the media stream through the MediaServer the routing of the RTP packets through the IP networks changes! ◊ If the QoS-problem disappears when forced via MediaServer and reappears when switched back then this is a strong indication that something is wrong in the direct IP routing path between the customer/user and the peer device.

Localize "No-Way Connection" and Possible Actions

"No-Way Connection":

◊ No voice transmission, neither from A->B nor B->A

Knowhow background:

◊ May occur during commissioning of the customer connection for VoIP
◊ May occur when the telephony provider introduce now IP networks for new telephony users
◊ May occur when the Internet service provider or telephony provider modify the IP routing
◊ Customer firewall policies block IP range or UDP port range
◊ The peer devices negotiate not the same codec
◊ May occur when IP devices are defect
◊ User device defect

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			Assumption: The problem reporting user/customer shall be the A leg. **1st:** Check if the negotiated "codec" are correct for both peers. ◊ Check in the SDP information if the selected codec of the leg B side is in the offered list of leg A

- The B side codec must be within the codec list of side A
  Possible problems: ouw-ing, No-Way or One-Way connection

Actions:
If the B side codec is not within the codec list of side A then check the configuration of the peer device B.
Check the configuration of the device A why the codec of side B is not in its list.
Consider a firmware upgrade of either device!

2nd: Check if the negotiated IP address and UDP ports are not blocked by any firewall.

◊ Check in the SDP information if the displayed IP addresses and UDP ports of the leg A and B are not blocked by any firewall
Possible problems: No-Way or One-Way connection

Actions:
Adjust the customers firewall policies
Adjust the usable UDP port range in the customer peer device

3rd: Check the `"rtp data"` records if the RTP transfer from and to the user/customer is not working:

◊ Are there are no (or few) received packets from the user?
◊ Are there packets sent toward the user?
If `Leg A "rec"=0` and `Leg A "snt">0`: no packets received from A but packets were sent toward A.

Actions:
If there is a gateway on the user/customer side which is provided from the Telephony provider then check the correct working of this gateway.
The user or customer IT responsible must check if the Internet or access to the Telephony network is ok.
The user or customer IT responsible must check if the access router or Firewall are ok (no blocking policies, VoIP ALG off, ...).
The user or customer IT responsible must check if its IT infrastructure is ok (no faulty IP switches, routers, ...).
The user or customer PBX responsible must check if the ISDN- or IP-PBX is working correctly.
The user must check if the telephone device is ok.

4th: Check the `"rtp data"` records if the RTP transfer from and to the PSTN is not working:

◊ Are there are no (or few) received packets from the PSTN?
◊ Are there packets sent toward the PSTN?
If `Leg B "rec"=0` and `Leg B "snt">0`: no packets received from B but packets were sent toward B

Actions:
The supporter must inform immediately the telephony provider support or IT emergency organization.

5th: Check the `"rtp data"` records if the RTP handling in the VoIP Switch MediaServer is not working:

◊ Are there are packets received from the PSTN but not sent toward the user?
If `Leg B "rec">0` and `Leg A "snt"=0`: packets from B received but no packets sent toward A
◊ Are there are packets received from the user but not sent toward the PSTN?
If `Leg A "rec">0` and `Leg B "snt"=0`: packets from A received but no packets sent toward B

Actions:
The supporter must inform immediately the telephony provider support!

Equipment			

Localize "One-Way Connection `A->B`" and Possible Actions

"One-Way Connection `A->B`":

◊ B hears A but A doesn't hear B

Knowhow background:

◊ May occur during commissioning of the customer connection for VoIP
◊ May occur when the telephony provider introduce now IP networks for new telephony users
◊ May occur when the Internet service provider or telephony provider modify the IP routing
◊ Customer firewall policies block IP range or UDP port range
◊ The peer devices negotiate not the same codec
◊ May occur when IP devices are defect
◊ User device defect

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			Assumption: The problem reporting user/customer shall be the A leg. **1st:** Check if the negotiated "codec" are correct for both peers. ◊ Check in the SDP information if the selected codec of the leg B side is in the offered list of leg A • The B side codec must be within the codec list of side A Possible problems: ouw-ing, No-Way or One-Way connection Actions: If the B side codec is not within the codec list of side A then check the configuration of the peer device B. Check the configuration of the device A why the codec of side B is not in its list. Consider a firmware upgrade of either device! **2nd:** Check if the negotiated IP address and UDP ports are not blocked by any firewall. ◊ Check in the SDP information if the displayed IP addresses and UDP ports of the leg A and B are not blocked by any firewall Possible problems: No-Way or One-Way connection Actions: Adjust the customers firewall policies Adjust the usable UDP port range in the customer peer device **3rd:** Check the `"rtp data"` records if the RTP transfer from the PSTN is not working: ◊ Are there are no (or few) received packets from the PSTN? If `Leg B "rec"=0`: no packets received from the PSTN. Actions: The supporter must inform immediately the telephony provider support or IT emergency organization.

			4th: Check the `"rtp data"` records if the RTP transfer to the user/customer is working: ◊ Are there are received packets from the PSTN and sent toward the user? If `Leg B "rec">0` and `Leg A "snt">0`: packets were received from the PSTN and sent toward the user. Actions: If there is a gateway on the user/customer side which is provided from the Telephony provider then check the correct working of this gateway. The user or customer IT responsible must check if the Internet or access to the Telephony network is ok. The user or customer IT responsible must check if the access router or Firewall are ok (no blocking policies, VoIP ALG off, ...). The user or customer IT responsible must check if its IT infrastructure is ok (no faulty IP switches, routers, ...). The user or customer PBX responsible must check if the ISDN- or IP-PBX is working correctly. The user must check if the telephone device is ok. **5th:** Check the `"rtp data"` records if the RTP handling in the VoIP Switch MediaServer is not working: ◊ Are there are packets received from the PSTN but not sent toward the user? If `Leg B "rec">0` and `Leg A "snt"=0`: packets were received from the PSTN but not sent toward the user Actions: The supporter must inform immediately the telephony provider support!
Equipment			

Localize "One-Way Connection `B->A`" and Possible Actions

"One-Way Connection `B->A`":

◊ A hears B but B doesn't hear A

Knowhow background:

◊ May occur during commissioning of the customer connection for VoIP
◊ May occur when the telephony provider introduce now IP networks for new telephony users
◊ May occur when the Internet service provider or telephony provider modify the IP routing
◊ Customer firewall policies block IP range or UDP port range
◊ The peer devices negotiate not the same codec
◊ May occur when IP devices are defect
◊ User device defect

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			Assumption: The problem reporting user/customer shall be the A leg. **1st:** Check if the negotiated "codec" are correct for both peers. ◊ Check in the SDP information if the selected codec of the leg B side is in the offered list of leg A • The B side codec must be within the codec list of side A

Possible problems: ouw-ing, No-Way or One-Way connection

Actions:
If the B side codec is not within the codec list of side A then check the configuration of the peer device B.
Check the configuration of the device A why the codec of side B is not in its list.
Consider a firmware upgrade of either device!

2nd: Check if the negotiated IP address and UDP ports are not blocked by any firewall.

◊ Check in the SDP information if the displayed IP addresses and UDP ports of the leg A and B are not blocked by any firewall

Possible problems: No-Way or One-Way connection

Actions:
Adjust the customers firewall policies
Adjust the usable UDP port range in the customer peer device

3rd: Check the `"rtp data"` records if the RTP transfer from the user is not working:

◊ Are there are no (or few) received packets from the user?
If `Leg A "rec"=0`: no packets received from the user

Actions:
If there is a gateway on the user/customer side which is provided from the Telephony provider then check the correct working of this gateway.
The user or customer IT responsible must check if the Internet or access to the Telephony network is ok.
The user or customer IT responsible must check if the access router or Firewall are ok (no blocking policies, VoIP ALG off, ...).
The user or customer IT responsible must check if its IT infrastructure is ok (no faulty IP switches, routers, ...).
The user or customer PBX responsible must check if the ISDN- or IP-PBX is working correctly.
The user must check if the telephone device is ok.

4th: Check the `"rtp data"` records if the RTP transfer to the PSTN is not working:

◊ Are there packets sent toward the PSTN?
If `Leg A "rec">0` and `Leg B "snt">0`: packets were received from the user but not sent toward the PSTN

Actions:
The supporter must inform immediately the telephony provider support or IT emergency organization.

5th: Check the `"rtp data"` records if the RTP handling in the VoIP Switch MediaServer is not working:

◊ Are there are packets received from the user but not sent toward the PSTN?
If `Leg A "snt">0` and `Leg B "snt"=0`,

Actions:
The supporter must inform immediately the telephony provider support!

Equipment

Localize "Glitch Connection" and Possible Actions

"Glitch Connection":

◊ The voice transmission from A->B and B->A is disturbed

Knowhow background:

◊ May occur when the customers Intranet is not optimized for VoIP
◊ The peer devices negotiate not the same codec
◊ May occur when IP devices are defect
◊ User device defect

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			Assumption: The problem reporting user/customer shall be the A leg. **1st:** Check if the negotiated "codec" are correct for both peers. ◊ Check in the SDP information if the selected codec of the leg B side is in the offered list of leg A: The B side codec must be within the codec list of side A Possible problems: ouw-ing, No-Way or One-Way connection Actions: If the B side codec is not within the codec list of side A then check the configuration of the peer device B. Check the configuration of the device A why the codec of side B is not in its list. Consider a firmware upgrade of either device! **2nd:** Check the `"rtcp data"` and `"rtp data"` records from and to the user/customer: ◊ Check the `"rtcp data"` (statistical data delivered from the device): `"lost>0"`: Device A claimed not to receive all packets Possible problems: crackle, short interruptions `"jitter>0"`: Device A claimed to receive packets delayed or wavering (if the jitter values are different then wavering) Possible problems: crackle, short interruptions, echo, ouw-ing ◊ Check the `"rtp data"` (statistical data from the VoIP Switch): `"rec"` not equal `"snt"`: The numbers of received `"rec"` and sent `"snt"` must be more or less equal since the last report. Possible problems: crackle, short interruptions `"cpkl>0.1"`: The cumulated packet loss `"cpkl"` should be smaller than `"<0.1"`. Possible problems: crackle, bigger interruptions Actions: If there is a gateway on the user/customer side which is provided from the Telephony provider then check the correct working of this gateway. The user or customer IT responsible must check if the Internet or access to the Telephony network is ok. The user or customer IT responsible must check if its IT infrastructure is ok (no faulty IP switches, routers, ...). The user or customer PBX responsible must check if IP-PBX is working correctly. The user must check if the telephone device is ok. **3rd:** Check the `"rtcp data"` and `"rtp data"`records from and to the PSTN is not working: ◊ Check the `"rtcp data"` (statistical data delivered from the device): `"lost>0"`: Device B claimed not to receive all packets.

			Possible problems: crackle, short interruptions "jitter>0": Device B claimed to receive packets delayed or wavering (if the jitter values are different then wavering). Possible problems: crackle, short interruption, echo, ouw-ing ◊ Check the "rtp data" (statistical data from the VoIP Switch): "rec" not equal "snt": The numbers of received "rec" and sent "snt" must be more or less equal since the last report Possible problems: crackle, short interruption "cpkl>0.1": The cumulated packet loss "cpkl" should be smaller than "<0.1" Possible problems: crackle, bigger interruptions Actions: If there are **no** similar problems in the big picture then the problem lies presumably in the network of side B. If there are **similar** problems in the big picture then the supporter must inform immediately the telephony provider support or IT emergency organization.
Equipment			

Localize "Glitch Connection A->B" and Possible Actions

"Glitch Connection A->B":

◊ The voice transmission from A->B is disturbed. B claims to hear A with bad quality.

Knowhow background:

◊ May occur when the customers Intranet is not optimized for VoIP
◊ May occur when IP devices are defect
◊ User device defect

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			Assumption: The problem reporting user/customer shall be the A leg. **1st:** Check if the negotiated "codec" are correct for both peers. ◊ Check in the SDP information if the selected codec of the leg B side is in the offered list of leg A: The B side codec must be within the codec list of side A Possible problems: ouw-ing, No-Way or One-Way connection Actions: If the B side codec is not within the codec list of side A then check the configuration of the peer device B. Check the configuration of the device A why the codec of side B is not in its list. Consider a firmware upgrade of either device! **2nd:** Check the "rtp data" records if the RTP transfer to the PSTN is correctly working: ◊ Check the "rtp data" (statistical data from the VoIP Switch) toward the PSTN: Leg A "rec" not equal Leg B "snt": The numbers of received "rec" and sent "snt" must be more or less equal since the last report. Possible problems: crackle, short interruptions Actions:

			If the received packets are very different to the sent ones then the supporter must inform immediately the telephony provider support or IT emergency organization. If the received packets are quite equal to the sent ones then the problem must be on the B side
Equipment			

Localize "Glitch Connection `B->A`" and Possible Actions

"Glitch Connection `B->A`":

◊ The voice transmission from B->A is disturbed. A claims to hear B with bad quality.

Knowhow background:

◊ May occur when the customers Intranet is not optimized for VoIP
◊ May occur when IP devices are defect
◊ User device defect

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			Assumption: The problem reporting user/customer shall be the A leg. **1st:** Check if the negotiated "codec" are correct for both peers. ◊ Check in the SDP information if the selected codec of the leg B side is in the offered list of leg A: The B side codec must be within the codec list of side A Possible problems: ouw-ing, No-Way or One-Way connection Actions: If the B side codec is not within the codec list of side A then check the configuration of the peer device B. Check the configuration of the device A why the codec of side B is not in its list. Consider a firmware upgrade of either device! **2nd:** Check the `"rtp data"` records if the RTP transfer to the user/customer is correctly working: ◊ Check the `"rtp data"` (statistical data from the VoIP Switch) toward the user/customer: `Leg B "rec"` not equal `Leg A "snt"`: The numbers of received `"rec"` and sent `"snt"` must be more or less equal since the last report. Possible problems: crackle, short interruptions Actions: If the received packets are very different to the sent ones then the supporter must inform immediately the telephony provider support or IT emergency organization. **3rd:** Check the `"rtcp data"` records from the user/customer: ◊ Check the `"rtcp data"` (statistical data delivered from the device): `"lost>0"`: Device A claimed not to receive all packets Possible problems: crackle, short interruptions `"jitter>0"`: Device A claimed to receive packets delayed or wavering (if the jitter values are different then wavering)

			Possible problems: crackle, short interruptions, echo, ouw-ing Actions: If there is a gateway on the user/customer side which is provided from the Telephony provider then check the correct working of this gateway. The user or customer IT responsible must check if the Internet or access to the Telephony network is ok. The user or customer IT responsible must check if its IT infrastructure is ok (no faulty IP switches, routers, ...). The user or customer PBX responsible must check if IP-PBX is working correctly. The user must check if the telephone device is ok.
Equipment			

Solve "Voice Glitches with ISDN-PBX" Problem

This problem type covers the following erroneous conditions:

◊ Bad speech quality in an ISDN-PBX environment
◊ Glitches in the voice transmission, it "clicks"

ISDN-PBX environment usually provide an excellent voice quality. In an VoIP environment this excellent voice quality can be only maintained if the ISDN-PBX can synchronize with high precision clock source.

1 Step: Check the ISDN reference clock

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			
Equipment	Checks: ◊ Does the ISDN-PBX take its clock reference from a high precision clock? Actions: Make sure the ISDN-PBX takes its reference clock from a high precision source. Use an ISDN-Gateway which provides a high precision clock.		

Solve "Special Telephony" Problem

Solve "FAX Transmission" Problem

This problem type covers the following erroneous conditions:

◊ FAX transmission doesn't start
◊ FAX transmission is dropped
◊ The transmitted document is incomplete

Note	If the FAX is connected to an IP-PBX then FAX problems must be solved with the responsible of the IP-PBX.

In a VoIP environment FAX no longer achieve the same degree of reliability as before in an analogue or ISDN one. The FAX reliability depends on various factors such as the type of device, device settings and the way the device is connected to the IP network. It depends also on the quality of the transmitter and receiver of the peer FAX devices. Getting all these factors together a transmission may not even start or dropped unexpectedly. The transmitted documents may be incomplete.

The users must expect increasing difficulties in the future, especially for international transmissions.

1 Step: Check the FAX device configuration

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			
Equipment	Check: ◊ the configuration of the FAX device Actions: Adjust the FAX device configuration: • Reduce the transmission speed to max. 9600bds. • Switch OFF the error correction as e.g. EMC • If the device offers a "VoIP mode" then experiment with it and check if the results are better.		

2 Step: Check the FAX transmission configuration of the gateway

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			
Equipment	Depending on the quality of the IP network the supporter and/or administrator of the gateway can experiment with the FAX transmission protocol of the gateway device. Checks: ◊ Check with the Telephony provider if there are recommendations or directives which type of FAX transmission protocols are to use, e.g.: · In band transmission with codec G.711alaw or G.711ulaw · Out band transmission with T.38 ◊ Check the configuration of the gateway device Actions: If the user/customer has a good quality IP network and the Telephone provider allows it then try: • "In band transmission with codec G.711" If the user/customer has a lower quality IP network and the Telephone provider allows it then try: • "Out band transmission with T.38"		

Solve "DECT Multi-Cell with ISDN-PBX" Problem

This problem type covers the following erroneous conditions:

◊ Hand over from cell to cell is not working
◊ Bad speech quality

Note	If the DECT Multi-Cell system is connected to an IP-PBX then DECT problems must be solved with the responsible of the IP-PBX.

DECT-Multi-Cell systems connected to an ISDN-PBX which is working with in a VoIP environment experience special issues. Most issues are interconnected with accuracy of the synchronization clock of the ISDN-PBX. If this synchronization clock is not especially precise then the reference clock of the DECT-Multi-Cell system will have problems as described above.

1 Step: Check the ISDN reference clock

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			
Equipment	Checks: ◊ Does the ISDN-PBX take its clock reference from a high precision clock? Actions: Make sure the ISDN-PBX takes its reference clock from a high precision source. Use an ISDN-Gateway which provides a high precision clock.		

p>

Via ConfigCenter check the users status and account configuration on the VoIP Switch:

1. Check "Validity":
   Check if the user account and its addresses are existing and "valid".
2. Check the telephone number registration status.
   If there is no registration you can proceed directly with → The user device is not registered ...
3. Check "TopStop":
   Check if a TopStop in the account or address prevents the user from doing outgoing calls.
4. Check "RuleSet":
   Check if a selected RuleSet in the account or address configuration prevents the user from doing outgoing or receiving calls.
5. Check "Call Forwards" or "Call Rejecting":
   Check if a "Call Forwards" or "Call Rejecting" in the account prevents the user from doing outgoing or receiving calls.
6. Check "Call Data":
   Consult the "Call Data" for the last connection attempts and connections longer than 2min of the user.

Step 3: Evaluate the User's VoIP Setup

For questioning and analyzing the user's problem it is necessary that the supporter is aware of the VoIP setup of the user.

The experienced supporter knows of the user's VoIP setup after the cross check . If not here the supporter finds the most implemented VoIP setup's.

VoIP Setup: Residential

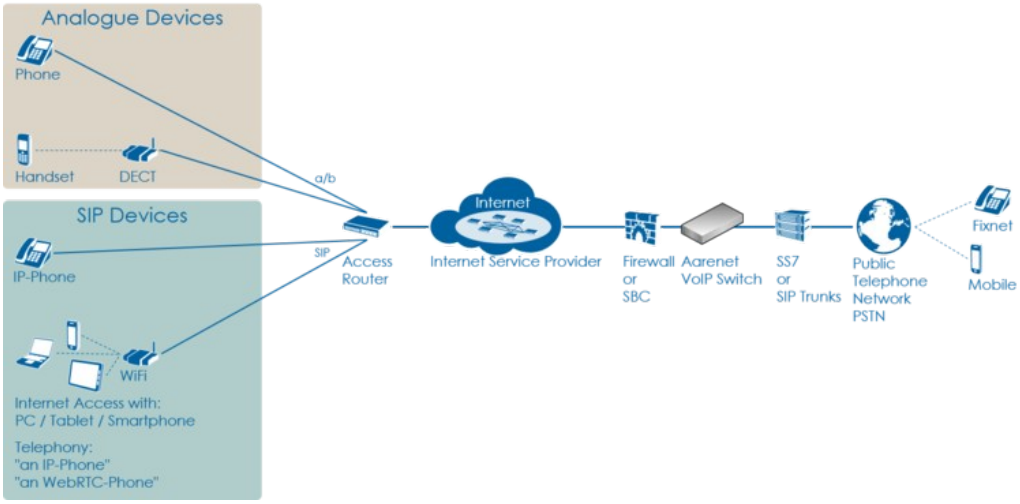
Characteristics:

◊ Private household
◊ Single or few telephone number
◊ Each telephone number registers individually

Most common problems:

◊ Account or telephone number blocked on the VoIP switch
◊ Telephone number not correctly ported to the telephony provider
◊ Telephone not correctly configured
◊ Telephone, cables defect
◊ Internet access fails

Overview VoIP Setup:



VoIP Setup: Legacy ISDN PBX

Characteristics:

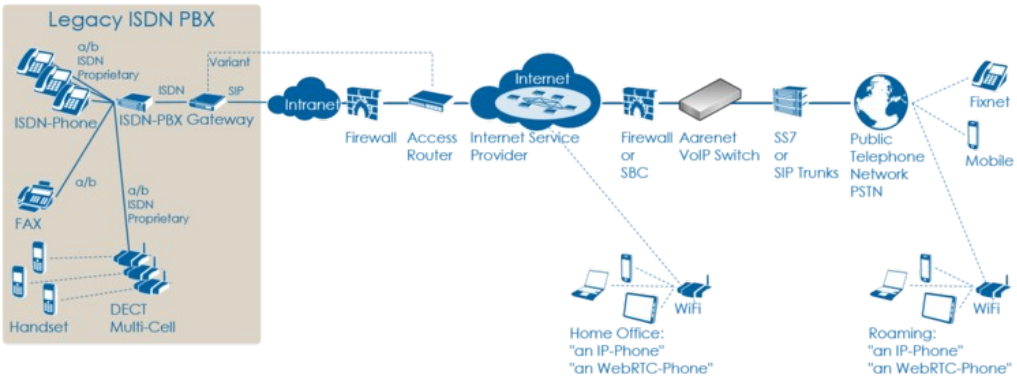
◊ Company PBX
◊ The ISDN PBX is connected via BRI or PRI to an ISDN-SIP Gateway
◊ One or more telephone number ranges
◊ The telephone numbers are registered via a main number
◊ The telephone number of incoming calls are signaled with only a few digits

Most common problems:

◊ Account or telephone numbers blocked on the VoIP switch
◊ Telephone number ranges not correctly ported to the telephony provider
◊ Telephone number ranges not completely configured on the VoIP Switch
◊ Wrong incoming telephone number signaling

◊ Internet access fails
◊ The company Firewall VoIP ALG interferes with the SIP signaling or needed IP ports are blocked.
◊ QoS problems for speech, Fax, DECT

Overview VoIP Setup:



VoIP Setup: IP PBX

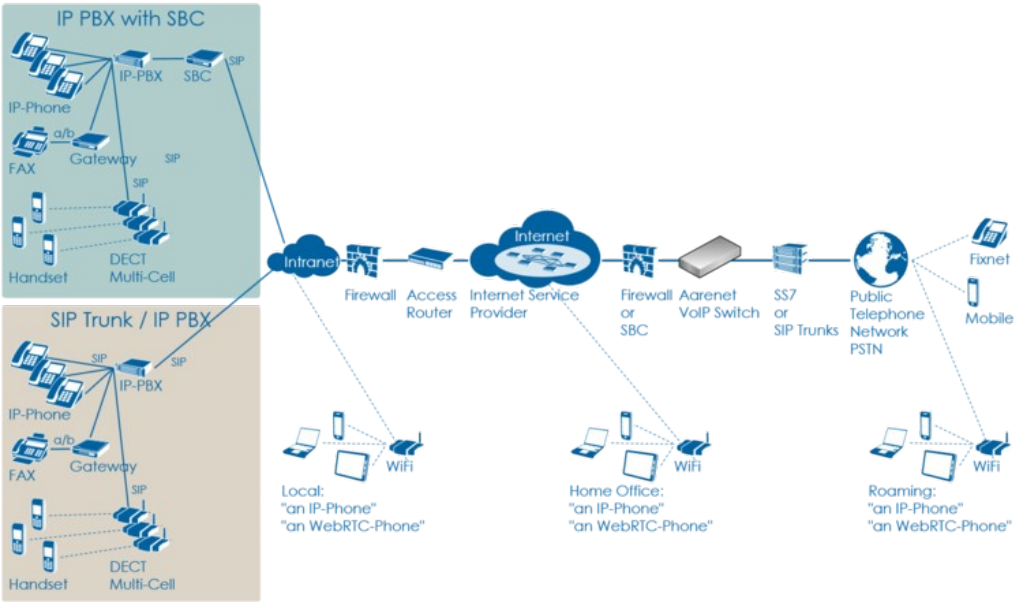
Characteristics:

◊ Company PBX
◊ The IP PBX is connected directly or via SBC to the VoIP Switch
◊ One or more telephone number ranges
◊ The telephone numbers are registered via a main number

Most common problems:

◊ Account or telephone numbers blocked on the VoIP switch
◊ Telephone number ranges not correctly ported to the telephony provider
◊ Telephone number ranges not completely configured on the VoIP Switch
◊ The company Firewall and/or SBC VoIP ALG interferes with the SIP signaling or needed IP ports are blocked.
◊ Internet access fails
◊ QoS problems for speech

Overview VoIP Setup:



VoIP Setup: vPBX

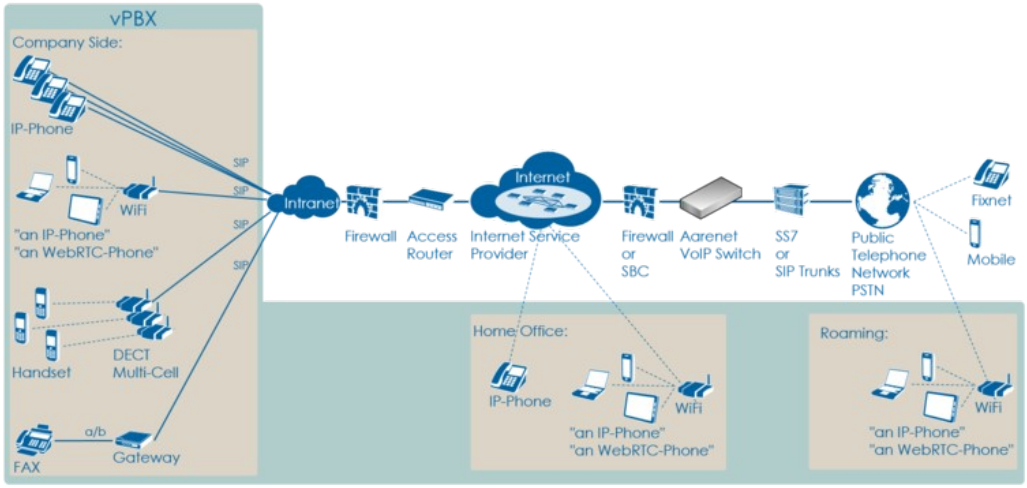
Characteristics:

◊ Company PBX
◊ The IP Phones are connected directly to the VoIP Switch
◊ One or more telephone number ranges

Most common problems:

◊ Public account and/or public telephone numbers blocked on the VoIP switch
◊ Public telephone number ranges not correctly ported to the telephony provider
◊ Telephone number ranges not completely configured on the VoIP Switch
◊ Private account and/or private telephone numbers blocked on the VoIP switch
◊ Provisioning of the SIP devices out of the AdminCenter
◊ The company or home office Firewalls and/or SBCs VoIP policies or ALG interferes with the SIP signaling or needed IP ports are blocked.
◊ Company/home office Internet access fails
◊ QoS problems for speech, FAX, DECT

Overview VoIP Setup:



Step 4: Check the "Big Picture"

At this point the supporter should get aware if the problem is limited to this user or if it could be large scale problem within the VoIP System.

If the supporter suspects a large scale problem, due to a great amount of the same ore similar user complains then he should contact the telephony provider support or emergency organization.

If the supporter has enough privileges he can check:

1. The VoIP Switch component status
   This will show if the VoIP Switch itself has a problem.
2. The VoIP System monitor
   Here you can check if:
   · The registrations dropped in a large scale
   · The calls dropped in a large scale
   · The IP connectivity somewhere in the VoIP system failed

At any rate the supporter **must inform** the VoIP system administrator!

Step 5: Solve the Customer Problem

Solve "Device / Network / Configuration / Registration" Problems

This problem type covers the following erroneous conditions:

◊ The device doesn't start
◊ The device doesn't integrate into the IP network
◊ The device is not correctly configured
◊ The device doesn't register at the VoIP Switch

Note	If the device is connected to an IP-PBX then these problems must be solved with the responsible of the IP-PBX.

Solve "Device Hardware & Firmware" Problem

1 Step: Is the device powered on, not defect?

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			
Equipment	Check if the device correctly powered and shows basic activity? ◊ Is the power cable correctly plugged in? ◊ Is the power cable not defect? ◊ Does the device show power on indication, e.g. display on, LED on? Actions: Replace the power cable Replace defect device if the powering is ok but no working indication is displayed		

Warning	Defect power cables must be replaced! Faulty power cables can be life-threatening!

2 Step: Is the device connected to the IP network?

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			

Equipment	Is the device correctly connected to the IP network? ◊ Is the patch cable correctly plugged in? ◊ Is the patch cable not defect? ◊ Are there LED flashing or glow next to the network plug on the device or at the peer device (access router, IP switch)? Actions: Replace the patch cable Plug in the patch cable at a different port at the peer device (access router, IP switch) Replace defect device if the patch cable and peer port is ok but no working indication is displayed at the device port.		

3 Step: Has the device a reasonable firmware loaded?

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			
Equipment	Has the device a reasonable firmware loaded? ◊ The user must check the loaded firmware Actions: Replace the firmware if outdated or important bugs are fixed		

Solve "Device Network" Problem

1 Step: Has the device an IP address and can access the Internet?

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer	Has the device got an IP address? ◊ Check on the device if it has received an IP address, e.g. via display or maintenance GUI. Actions: ◊ If no IP address was assigned the user must: ◊ Check if the device is really connected to the IP network! ◊ Check if the device is configured with a fixed IP address! If it has a fixed IP does it match with the IP subnet? Is the default GW and DNS entry configured? ◊ Check if the device is configured to use DHCP! if the DHCP service in the IP network is running If the user cannot check the DHCP service he must contact the company IT responsible or the responsible of maintaining the access router. Has the device access toward the VoIp Switch? ◊ Check if the device makes contact with the VoIP Switch via Internet or any private IP Link.		

	Actions: ◊ If the device shall connect via the Internet: Connect a PC to the Ethernet port where the device usually is connected and try to connect to any public Web site. ◊ If the device shall connect via an private network check with the IT responsible if the access to the VoIP Switch is guaranteed.		
Equipment			

Solve "Registration" Problem

1 Step: Review the account and telephone number configuration

	Customer	Internet ISP	Telephony Provider
Telephony			Check via ConfigCenter: ◊ Does the user account exist and is it "valid"? ◊ Does the telephone number exist and is it "valid"? Actions: ◊ Check why the account, telephone number doesn't exist or is disabled activate them if allowed.
Data Transfer			
Equipment			

2 Step: Where REGISTER messages received from the device on the VoIP Switch?

	Customer	Internet ISP	Telephony Provider
Telephony			In the "Support Log" search for the device registration in the present and past time. 1. Set the "Support Log" filters · Insert at "Text" the account name · Insert at "From" - "Until" a reasonable time span where registrations could be expected · Select the category: "Registration" 2. Start the search and find out according the results what is going wrong. 3. If needed repeat the search with the telephone number in the "Text" or other time spans ◊ Check the log results → see below Actions: ◊ → see below
Data Transfer			
Equipment			

Failed registrations due to disabled account or address:

	2017-09-15-07:56:49.553 Registration failed, disabled account aan1-00093 tried to register number 0449980010

Actions:

◊ Check why the account is disabled and activate if allowed.

Failed registrations due to wrong SIP credentials:

	2017-09-15-08:05:38.117 Registration failed, invalid credentials for account acc-01 2017-09-15-08:05:39.112 Registration failed, unknown username 'myusername' tried to register '0123456789' 2017-09-15-08:05:38.377 Registration failed, unknown number '0987654321' tried to register for account acc-01

Actions:

◊ The user must manually adjust the SIP credentials on the device
◊ The user must re-configure the device via AdminCenter

The device didn't refresh its registration:

	2017-09-15-07:59:00.862 RegID989961 ended for 0987654321 ip=111.111.111.111:65398 ua=my-device v1.0

Actions:

◊ Order the user to check if the device is really on-line!
◊ Order the user to check if the device is defect? powered on? patch? IP address? see below

For information a successful registration:

	2017-09-15-07:59:30.383 RegID989965 started for 0987654321 ip=111.111.111.111:65398 ua=my-device v1.0

Hint:
The supporter might try to find REGISTER messages from the device in the "Trace" . This gives the certainty that the message was received by the VoIP switch. The supporter can filter for the telephone number. If the IP address is needed then the customer must be able to tell or evaluate it, e.g.:

```
https://www.whatismyip.com/
```

3 Step: Is the device correctly configured for registration??

	Customer	Internet ISP	Telephony Provider
Telephony	For a manually configured device, check that the device has the correct configuration for: ◊ Telephone number ◊ SIP credentials ◊ VoIP Switch domain configuration Actions: The user must manually check the device configuration and if needed adjust its configuration of the telephone number, SIP credentials and VoIP Switch domain for registration For a automatically via AdminCenter configured device check that: ◊ the selected device type in the AdminCenter is identical to the physical one. Actions:		

	If not the same type then the user must re-configure the device via AdminCenter For a automatically via AdminCenter configured device check that: ◊ the user device has downloaded its configuration. Actions: If the configuration is not downloaded then it must be checked if the device: ◊ has got an IP address in the local IP network ◊ has access to the Internet ◊ has access to the configuration download of the Telephony Provider. By default this is the IP address of the VoIP Switch domain and uses the protocol HTTPS on TCP port 443. Check with the Telephone provider or via ConfigCenter > Menu "System" > "Zone Profiles" The user must check if the Firewall, SBC or Access Router doesn't block HTTPS traffic to and from the configuration download of the Telephony Provider		
Data Transfer			
Equipment			

Solve "Connection" Problems

This problem type covers the following erroneous conditions:

◊ Incoming or outgoing calls are not working
◊ Wrong called number
◊ Call supervision
◊ User device not registered
◊ User device not correct configured
◊ SIP signaling in general

Note	If the device is connected to an IP-PBX then these problems must be solved with the responsible of the IP-PBX.

1 Step: Review the account and telephone number configuration / registration?

	Customer	Internet ISP	Telephony Provider
Telephony			Do this check for the A and/or B telephone number if they are on-net numbers of the VoIP SWitch. Check via ConfigCenter: ◊ Does the telephone number exist? ◊ Is the telephone number valid? ◊ Is the user account valid? ◊ Is the telephone number correctly registered Actions: Check why the account, telephone number doesn't exist or is disabled and activate if allowed.

			Check why the device is not registered at the VoIP Switch
Data Transfer			
Equipment			

Hint:

◊ If the device is not registered outgoing calls might be working but NO incoming call will work.

2 Step: Was the called number correctly transmitted to the peer?

	Customer	Internet ISP	Telephony Provider
Telephony			Check via ConfigCenter: 1. In the "Call Data" search for the erroneous call: ♦ Set the "Call Data" filters: · Insert at "Time" a reasonable time span where the erroneous call is to be expected. · Set "Duration" to 00:00:00 · Insert at "Called Number" the called number 2. Start the search and identify the CDR of the erroneous call in the list If no CDR was found search for the "Calling Number"! 3. Open the identified CDR and get the trace of the call, click the Button [Trace] 4. Check if the called number in the "TO-Header" in all INVITE messages is correct: ♦ Is the called number correct? Often the users don't dial all digits or wrong digits or the configured number on a direct call key is incorrect. ♦ If the number is dialed correctly then it can be that the destination is not reachable. ♦ Outgoing calls from a vPBX can miss the public prefix 5. Check the peers call cancel reason: ♦ SIP Failure Responses Actions: Inform the user to dial the correct number. Try to reach the called number via an alternative telephone network, e.g. from a mobile telephone. Check with the support of the telephony provider why the called number is not reachable.
Data Transfer			
Equipment			

3 Step: What is the reason of an interrupted connection?

	Customer	Internet ISP	Telephony Provider
Telephony	Search in the "Call Data" for the erroneous call: 1. ♦ Set the "Call Data" filters: · Insert at "Time" a reasonable time span where the erroneous call is to be expected. · Set "Duration" to 00:00:00 · Insert at "Called Number" the called number 2. Start the search and identify the CDR of the erroneous call in the list If no CDR was found search for the "Calling Number"! Open the identified CDR and check for the release or reject reason:		

	◊ Was the connection released by a peer, A or B side? ◊ Check cancel reason in "State" (SIP Failure Responses) Actions: Inform the user about the release reason, e.g. his own device or the peer device released the call regularly (but probably nor expected). Check if a call was released due to call supervision: ◊ Variant 1: Session Timer was not refreshed: · Open the "Trace" of the connection and check if the connection was released due to missing RE-INVITE from the peers when the Session Timer run out. Actions: Inform the user that his device did not restart the Session Timer. The device configuration must be inspected an adjusted if needed. ◊ Variant 2: SIP INFO were not answered by the peer: · Open the "Trace" of the connection and check if the connection was released due to not answered INFO messages that were sent from the VoIP Switch toward the peers. If activated the INFO's are sent usually every 120sec. Actions: Inform the user that his device did not answered INFO messages. It must be checke with the support of the device manufacturer if the device doesn't send 200 ACK when an empty INFO message was received ("SIP ping"). ◊ Variant 3: Missing RTP packets between the peers: This type of problem is a difficult one and hard to check and solve! It must be handled like a QoS problem. Media transferring devices as the MediaServer of the VoIP Switch, SBCs, SS7-Gateways, SIP-Trunks supervise the media stream of RTP packets. If after a certain time no RTP packets are transferred in a connection then such an instance can release the call. Typically after 30secs a connection is released if no RTP streams are detected. ◊ · Open the "Media Trace" of the connection and check if there are remarkable differences between the amount of sent and received or lost RTP packets. Actions: ➔ See "Quality of Service QoS problem" below.		
Data Transfer			
Equipment			

Solve "Quality of Service QoS" QoS-Problems

Introduction to QoS-Problems

Note	In most cases, QoS-problems can only be found and solved by means of an exclusion procedure. It is paramount that the customer/user knows that QoS-problems are difficult to track down and to solve. It's nerve-wracking and it is time consuming. Solving QoS-problems often requires the cooperation and active co-testing from the customer/user with the support personnel! The active help of the customer/user is needed in most cases, e.g. by executing test connections.

The QoS-problem type covers the following erroneous conditions:

◊ No voice transmission in one or both directions from the beginning of the connection
◊ Bad voice quality during the connection

Naming and characteristics of QoS-problem:

One/No-Way Connection:

There is no speech transmission in one or both directions from beginning of the connection:

- Silence (Possible reason: Mostly due to no or blocked RTP data transmission)

Glitch Connection:

There is speech transmission but it is disturbed:

- Crackle, clicking (Possible reason: small packet loss, jitter)
- Short interruption (Possible reason: bigger packet loss)
- Ouw-ing (Possible reason: jitter, transcoding)
- Echo (Possible reason: jitter, big delay)

The source of the QoS-problems are all too often somewhere in the data transmission "D Data Transfer" layer (but sometimes they are surprisingly simple):

◊ The microphone or loadspeaker in the telephone handset defect
◊ Volume configuration in the telephone set wrong
◊ Telephone defect
◊ The company Intranet is not made ready for VoIP
◊ Any device in the "D - Data Transfer" layer

1st Step: Interview the User

1 Step: Interview the user carefully and identify the type of QoS-problem

Get all information from the user:

1. Occurs the the QoS-problem with all peers or just with the given B peer?
   Hint:
   If the problem occurs only with the B peer then this is a strong indication that something is wrong on the B side!
2. Is there no voice transmission, neither from `A->B` nor `B->A`?
   Type of QoS-problem: "No-Way Connection"
3. Is there voice transmission from `A->B` (B hears you) but none from `B->A` (you don't hear B)?
   Type of QoS-problem: "One-Way Connection `A->B`"
4. Is there voice transmission from `B->A` (you hear B) but none from `A->B` (B doesn't hear you)?
   Type of QoS-problem: "One-Way Connection `B->A`"
5. Are there during the connection crackle, clicking, short interruptions, uow-ing in the voice transmission for both sides?
   Type of QoS-problem: "Glitch Connection"
6. Are there during the connection crackle, clicking, short interruptions, uow-ing in the voice transmission from `A->B`?

Type of QoS-problem: "Glitch Connection A->B"

7. Are there during the connection crackle, clicking, short interruptions, uow-ing in the voice transmission from B->A?
   Type of QoS-problem: "Glitch Connection B->A"
8. Uses the user an ISDN or DECT telephone behind an ISDN-PBX? Does the user have sharp clicking glitches in a regular or irregular interval? Do experience all users behind this ISDN-PBX this clicking?
   Remember :
   This points to a synchronization problem of the ISDN-PBX!
9. Is one peer A or B a mobile user?
   Remember:
   Mobile networks often have QoS-problems on the wireless links between the base station and the mobile device!

Action:

Cross check the users information by checking the media transfer statistics of the affected connection, see "2 Step" below!

2nd Step: Localize the QoS-Problem

Note	It is very important that the supporter is aware of of the localization of the problem. QoS-problems in the range of the "2 Internet Service Provider ISP" or "3 Telephony Provider" will affect usually a lot of users immediately.

1 Step: Check the "Big Picture"

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer	Check with the ISP where the user is connected to if there are outages in: ◊ the ISP IP network Actions: Inform immediately the Telephone providers support or alarming organization. If yes, just wait until the outage is solved	Check with the ISP where the VoIP System is connected to if there are outages in: ◊ the ISP IP network ◊ relevant national and/or international IP network Actions: Inform immediately the Telephone providers support or alarming organization. If yes, just wait until the outage is solved	Check with the IT responsible of the IP network where the VoIP System is attached to: ◊ Are there known outages in the IP network where the VoIP System is attached to? ◊ Is there a large scale QoS-problem? ◊ Are users affected which are located: • in a certain private IP network of the telephony provider? • at a definable tenant? Actions: Inform immediately the Telephone providers support or alarming

			organization. If the VoIP System is located in a pure private IP network then contact immediately the IT responsible or IT emergency organization. If yes, just wait until the outage is solved
Equipment			

2 Step: Identify the disturbed transmission direction from the VoIP Switch's view

This identification bases upon the VoIP System setting that all media streams are routed via the MediaServer of the VoIP Switch. The MediaServer collects statistic information about all media stream that are routed through it. These statistics can help to identify the source of the QoS-problem.

Search in the "Call Data" the CDR of the erroneous call:

1. ♦ Set the "Call Data" filters:
   · Insert at "Time" a reasonable time span where the erroneous call is to be expected.
   · Set "Duration" to 00:00:00
   · Insert at "Called Number" the called number
2. Start the search and identify the CDR of the erroneous call in the list
   If no CDR was found search for the "Calling Number"!
3. Open the identified CDR
4. Get the RTP statistics of this connection, click Button [Media Trace]
   If there are no data in the "Media Trace" contained then the media stream is not routed via the MediaServer of the VoIP Switch. See below how to force the routing via the MediaServer.
   Depending of the identified QoS-problem type analyze the RTP statistics detail, see below

If the media stream are not routed by default via MediaServer the supporter can force it for an account via the ConfigCenter:

→ Menu "Account"

→ Select the customers account

→ Tab "Advanced"

→ Set "Use always MediaServer" to `"Yes"`

Note	◊ By forcing the media stream through the MediaServer the routing of the RTP packets through the IP networks changes! ◊ If the QoS-problem disappears when forced via MediaServer and reappears when switched back then this is a strong indication that something is wrong in the direct IP routing path between the customer/user and the peer device.

Localize "No-Way Connection" and Possible Actions

"No-Way Connection":

◊ No voice transmission, neither from A->B nor B->A

Knowhow background:

◊ May occur during commissioning of the customer connection for VoIP
◊ May occur when the telephony provider introduce now IP networks for new telephony users
◊ May occur when the Internet service provider or telephony provider modify the IP routing
◊ Customer firewall policies block IP range or UDP port range
◊ The peer devices negotiate not the same codec
◊ May occur when IP devices are defect
◊ User device defect

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			Assumption: The problem reporting user/customer shall be the A leg. **1st:** Check if the negotiated "codec" are correct for both peers. ◊ Check in the SDP information if the selected codec of the leg B side is in the offered list of leg A • The B side codec must be within the codec list of side A Possible problems: ouw-ing, No-Way or One-Way connection Actions: If the B side codec is not within the codec list of side A then check the configuration of the peer device B. Check the configuration of the device A why the codec of side B is not in its list. Consider a firmware upgrade of either device! **2nd:** Check if the negotiated IP address and UDP ports are not blocked by any firewall. ◊ Check in the SDP information if the displayed IP addresses and UDP ports of the leg A and B are not blocked by any firewall Possible problems: No-Way or One-Way connection Actions: Adjust the customers firewall policies Adjust the usable UDP port range in the customer peer device **3rd:** Check the `"rtp data"` records if the RTP transfer from and to the user/customer is not working: ◊ Are there are no (or few) received packets from the user? ◊ Are there packets sent toward the user? If `Leg A "rec"=0` and `Leg A "snt">0`: no packets received from A but packets were sent toward A. Actions: If there is a gateway on the user/customer side which is provided from the Telephony provider then check the correct working of this gateway. The user or customer IT responsible must check if the Internet or access to the Telephony network is ok. The user or customer IT responsible must check if the access router or Firewall are ok (no blocking policies, VoIP ALG off, ...).

			The user or customer IT responsible must check if its IT infrastructure is ok (no faulty IP switches, routers, ...). The user or customer PBX responsible must check if the ISDN- or IP-PBX is working correctly. The user must check if the telephone device is ok. **4th:** Check the `"rtp data"` records if the RTP transfer from and to the PSTN is not working: ◊ Are there are no (or few) received packets from the PSTN? ◊ Are there packets sent toward the PSTN? If `Leg B "rec"=0` and `Leg B "snt">0`: no packets received from B but packets were sent toward B Actions: The supporter must inform immediately the telephony provider support or IT emergency organization. **5th:** Check the `"rtp data"` records if the RTP handling in the VoIP Switch MediaServer is not working: ◊ Are there are packets received from the PSTN but not sent toward the user? If `Leg B "rec">0` and `Leg A "snt"=0`: packets from B received but no packets sent toward A ◊ Are there are packets received from the user but not sent toward the PSTN? If `Leg A "rec">0` and `Leg B "snt"=0`: packets from A received but no packets sent toward B Actions: The supporter must inform immediately the telephony provider support!
Equipment			

Localize "One-Way Connection `A->B`" and Possible Actions

"One-Way Connection `A->B`":

◊ B hears A but A doesn't hear B

Knowhow background:

◊ May occur during commissioning of the customer connection for VoIP
◊ May occur when the telephony provider introduce now IP networks for new telephony users
◊ May occur when the Internet service provider or telephony provider modify the IP routing
◊ Customer firewall policies block IP range or UDP port range
◊ The peer devices negotiate not the same codec
◊ May occur when IP devices are defect
◊ User device defect

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			Assumption: The problem reporting user/customer shall be the A leg. **1st:** Check if the negotiated "codec" are correct for both peers. ◊ Check in the SDP information if the selected codec of the leg B side is in the offered list of leg A • The B side codec must be within the codec list of side A

Possible problems: ouw-ing, No-Way or One-Way connection

Actions:
If the B side codec is not within the codec list of side A then check the configuration of the peer device B.
Check the configuration of the device A why the codec of side B is not in its list.
Consider a firmware upgrade of either device!

2nd: Check if the negotiated IP address and UDP ports are not blocked by any firewall.

◊ Check in the SDP information if the displayed IP addresses and UDP ports of the leg A and B are not blocked by any firewall

Possible problems: No-Way or One-Way connection

Actions:
Adjust the customers firewall policies
Adjust the usable UDP port range in the customer peer device

3rd: Check the `"rtp data"` records if the RTP transfer from the PSTN is not working:

◊ Are there are no (or few) received packets from the PSTN?
If `Leg B "rec"=0`: no packets received from the PSTN.

Actions:
The supporter must inform immediately the telephony provider support or IT emergency organization.

4th: Check the `"rtp data"` records if the RTP transfer to the user/customer is working:

◊ Are there are received packets from the PSTN and sent toward the user?
If `Leg B "rec">0` and `Leg A "snt">0`: packets were received from the PSTN and sent toward the user.

Actions:
If there is a gateway on the user/customer side which is provided from the Telephony provider then check the correct working of this gateway.
The user or customer IT responsible must check if the Internet or access to the Telephony network is ok.
The user or customer IT responsible must check if the access router or Firewall are ok (no blocking policies, VoIP ALG off, ...).
The user or customer IT responsible must check if its IT infrastructure is ok (no faulty IP switches, routers, ...).
The user or customer PBX responsible must check if the ISDN- or IP-PBX is working correctly.
The user must check if the telephone device is ok.

5th: Check the `"rtp data"` records if the RTP handling in the VoIP Switch MediaServer is not working:

◊ Are there are packets received from the PSTN but not sent toward the user?
If `Leg B "rec">0` and `Leg A "snt"=0`: packets were received from the PSTN but not sent toward the user

Actions:
The supporter must inform immediately the telephony provider support!

Equipment

Localize "One-Way Connection `B->A`" and Possible Actions

"One-Way Connection `B->A`":

◊ A hears B but B doesn't hear A

Knowhow background:

◊ May occur during commissioning of the customer connection for VoIP
◊ May occur when the telephony provider introduce now IP networks for new telephony users
◊ May occur when the Internet service provider or telephony provider modify the IP routing
◊ Customer firewall policies block IP range or UDP port range
◊ The peer devices negotiate not the same codec
◊ May occur when IP devices are defect
◊ User device defect

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			Assumption: The problem reporting user/customer shall be the A leg. **1st:** Check if the negotiated "codec" are correct for both peers. ◊ Check in the SDP information if the selected codec of the leg B side is in the offered list of leg A • The B side codec must be within the codec list of side A Possible problems: ouw-ing, No-Way or One-Way connection Actions: If the B side codec is not within the codec list of side A then check the configuration of the peer device B. Check the configuration of the device A why the codec of side B is not in its list. Consider a firmware upgrade of either device! **2nd:** Check if the negotiated IP address and UDP ports are not blocked by any firewall. ◊ Check in the SDP information if the displayed IP addresses and UDP ports of the leg A and B are not blocked by any firewall Possible problems: No-Way or One-Way connection Actions: Adjust the customers firewall policies Adjust the usable UDP port range in the customer peer device **3rd:** Check the `"rtp data"` records if the RTP transfer from the user is not working: ◊ Are there are no (or few) received packets from the user? If `Leg A "rec"=0`: no packets received from the user Actions: If there is a gateway on the user/customer side which is provided from the Telephony provider then check the correct working of this gateway. The user or customer IT responsible must check if the Internet or access to the Telephony network is ok. The user or customer IT responsible must check if the access router or Firewall are ok (no blocking policies, VoIP ALG off, ...). The user or customer IT responsible must check if its IT infrastructure is ok (no faulty IP switches, routers, ...). The user or customer PBX responsible must check if the ISDN- or IP-PBX is working correctly.

			The user must check if the telephone device is ok. **4th:** Check the `"rtp data"` records if the RTP transfer to the PSTN is not working: ◊ Are there packets sent toward the PSTN? If `Leg A "rec">0` and `Leg B "snt">0`: packets were received from the user but not sent toward the PSTN Actions: The supporter must inform immediately the telephony provider support or IT emergency organization. **5th:** Check the `"rtp data"` records if the RTP handling in the VoIP Switch MediaServer is not working: ◊ Are there are packets received from the user but not sent toward the PSTN? If `Leg A "snt">0` and `Leg B "snt"=0`, Actions: The supporter must inform immediately the telephony provider support!
Equipment			

Localize "Glitch Connection" and Possible Actions

"Glitch Connection":

◊ The voice transmission from A->B and B->A is disturbed

Knowhow background:

◊ May occur when the customers Intranet is not optimized for VoIP
◊ The peer devices negotiate not the same codec
◊ May occur when IP devices are defect
◊ User device defect

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			Assumption: The problem reporting user/customer shall be the A leg. **1st:** Check if the negotiated "codec" are correct for both peers. ◊ Check in the SDP information if the selected codec of the leg B side is in the offered list of leg A: The B side codec must be within the codec list of side A Possible problems: ouw-ing, No-Way or One-Way connection Actions: If the B side codec is not within the codec list of side A then check the configuration of the peer device B. Check the configuration of the device A why the codec of side B is not in its list. Consider a firmware upgrade of either device! **2nd:** Check the `"rtcp data"` and `"rtp data"` records from and to the user/customer: ◊ Check the `"rtcp data"` (statistical data delivered from the device):

			"lost>0": Device A claimed not to receive all packets Possible problems: crackle, short interruptions "jitter>0": Device A claimed to receive packets delayed or wavering (if the jitter values are different then wavering) Possible problems: crackle, short interruptions, echo, ouw-ing ◊ Check the "rtp data" (statistical data from the VoIP Switch): "rec" not equal "snt": The numbers of received "rec" and sent "snt" must be more or less equal since the last report. Possible problems: crackle, short interruptions "cpkl>0.1": The cumulated packet loss "cpkl" should be smaller than "<0.1". Possible problems: crackle, bigger interruptions Actions: If there is a gateway on the user/customer side which is provided from the Telephony provider then check the correct working of this gateway. The user or customer IT responsible must check if the Internet or access to the Telephony network is ok. The user or customer IT responsible must check if its IT infrastructure is ok (no faulty IP switches, routers, ...). The user or customer PBX responsible must check if IP-PBX is working correctly. The user must check if the telephone device is ok. **3rd:** Check the "rtcp data" and "rtp data"records from and to the PSTN is not working: ◊ Check the "rtcp data" (statistical data delivered from the device): "lost>0": Device B claimed not to receive all packets. Possible problems: crackle, short interruptions "jitter>0": Device B claimed to receive packets delayed or wavering (if the jitter values are different then wavering). Possible problems: crackle, short interruption, echo, ouw-ing ◊ Check the "rtp data" (statistical data from the VoIP Switch): "rec" not equal "snt": The numbers of received "rec" and sent "snt" must be more or less equal since the last report Possible problems: crackle, short interruption "cpkl>0.1": The cumulated packet loss "cpkl" should be smaller than "<0.1" Possible problems: crackle, bigger interruptions Actions: If there are **no** similar problems in the big picture then the problem lies presumably in the network of side B. If there are **similar** problems in the big picture then the supporter must inform immediately the telephony provider support or IT emergency organization.
Equipment			

Localize "Glitch Connection A->B" and Possible Actions

"Glitch Connection A->B":

◊ The voice transmission from A->B is disturbed. B claims to hear A with bad quality.

Knowhow background:

◊ May occur when the customers Intranet is not optimized for VoIP
◊ May occur when IP devices are defect
◊ User device defect

Customer	Internet ISP	Telephony Provider

Telephony			
Data Transfer			Assumption: The problem reporting user/customer shall be the A leg. **1st:** Check if the negotiated "codec" are correct for both peers. ◊ Check in the SDP information if the selected codec of the leg B side is in the offered list of leg A: The B side codec must be within the codec list of side A Possible problems: ouw-ing, No-Way or One-Way connection Actions: If the B side codec is not within the codec list of side A then check the configuration of the peer device B. Check the configuration of the device A why the codec of side B is not in its list. Consider a firmware upgrade of either device! **2nd:** Check the `"rtp data"` records if the RTP transfer to the PSTN is correctly working: ◊ Check the `"rtp data"` (statistical data from the VoIP Switch) toward the PSTN: `Leg A "rec"` not equal `Leg B "snt"`: The numbers of received `"rec"` and sent `"snt"` must be more or less equal since the last report. Possible problems: crackle, short interruptions Actions: If the received packets are very different to the sent ones then the supporter must inform immediately the telephony provider support or IT emergency organization. If the received packets are quite equal to the sent ones then the problem must be on the B side
Equipment			

Localize "Glitch Connection `B->A`" and Possible Actions

"Glitch Connection `B->A`":

◊ The voice transmission from B->A is disturbed. A claims to hear B with bad quality.

Knowhow background:

◊ May occur when the customers Intranet is not optimized for VoIP
◊ May occur when IP devices are defect
◊ User device defect

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			Assumption: The problem reporting user/customer shall be the A leg. **1st:** Check if the negotiated "codec" are correct for both peers. ◊ Check in the SDP information if the selected codec of the leg B side is in the offered list of leg A: The B side codec must be within the codec list of side A Possible problems: ouw-ing, No-Way or One-Way connection

			Actions: If the B side codec is not within the codec list of side A then check the configuration of the peer device B. Check the configuration of the device A why the codec of side B is not in its list. Consider a firmware upgrade of either device! **2nd:** Check the `"rtp data"` records if the RTP transfer to the user/customer is correctly working: ◊ Check the `"rtp data"` (statistical data from the VoIP Switch) toward the user/customer: `Leg B "rec"` not equal `Leg A "snt"`: The numbers of received `"rec"` and sent `"snt"` must be more or less equal since the last report. Possible problems: crackle, short interruptions Actions: If the received packets are very different to the sent ones then the supporter must inform immediately the telephony provider support or IT emergency organization. **3rd:** Check the `"rtcp data"` records from the user/customer: ◊ Check the `"rtcp data"` (statistical data delivered from the device): `"lost>0"`: Device A claimed not to receive all packets Possible problems: crackle, short interruptions `"jitter>0"`: Device A claimed to receive packets delayed or wavering (if the jitter values are different then wavering) Possible problems: crackle, short interruptions, echo, ouw-ing Actions: If there is a gateway on the user/customer side which is provided from the Telephony provider then check the correct working of this gateway. The user or customer IT responsible must check if the Internet or access to the Telephony network is ok. The user or customer IT responsible must check if its IT infrastructure is ok (no faulty IP switches, routers, ...). The user or customer PBX responsible must check if IP-PBX is working correctly. The user must check if the telephone device is ok.
Equipment			

Solve "Voice Glitches with ISDN-PBX" Problem

This problem type covers the following erroneous conditions:

◊ Bad speech quality in an ISDN-PBX environment
◊ Glitches in the voice transmission, it "clicks"

ISDN-PBX environment usually provide an excellent voice quality. In an VoIP environment this excellent voice quality can be only maintained if the ISDN-PBX can synchronize with high precision clock source.

1 Step: Check the ISDN reference clock

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			
Equipment	Checks:		

	◊ Does the ISDN-PBX take its clock reference from a high precision clock? Actions: Make sure the ISDN-PBX takes its reference clock from a high precision source. Use an ISDN-Gateway which provides a high precision clock.		

Solve "Special Telephony" Problem

Solve "FAX Transmission" Problem

This problem type covers the following erroneous conditions:

◊ FAX transmission doesn't start
◊ FAX transmission is dropped
◊ The transmitted document is incomplete

Note	If the FAX is connected to an IP-PBX then FAX problems must be solved with the responsible of the IP-PBX.

In a VoIP environment FAX no longer achieve the same degree of reliability as before in an analogue or ISDN one. The FAX reliability depends on various factors such as the type of device, device settings and the way the device is connected to the IP network. It depends also on the quality of the transmitter and receiver of the peer FAX devices. Getting all these factors together a transmission may not even start or dropped unexpectedly. The transmitted documents may be incomplete.

The users must expect increasing difficulties in the future, especially for international transmissions.

1 Step: Check the FAX device configuration

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			
Equipment	Check: ◊ the configuration of the FAX device Actions: Adjust the FAX device configuration: • Reduce the transmission speed to max. 9600bds. • Switch OFF the error correction as e.g. EMC • If the device offers a "VoIP mode" then experiment with it and check if the results are better.		

2 Step: Check the FAX transmission configuration of the gateway

	Customer	Internet ISP	Telephony Provider

Telephony			
Data Transfer			
Equipment	Depending on the quality of the IP network the supporter and/or administrator of the gateway can experiment with the FAX transmission protocol of the gateway device. Checks: ◊ Check with the Telephony provider if there are recommendations or directives which type of FAX transmission protocols are to use, e.g.: · In band transmission with codec G.711alaw or G.711ulaw · Out band transmission with T.38 ◊ Check the configuration of the gateway device Actions: If the user/customer has a good quality IP network and the Telephone provider allows it then try: • "In band transmission with codec G.711" If the user/customer has a lower quality IP network and the Telephone provider allows it then try: • "Out band transmission with T.38"		

Solve "DECT Multi-Cell with ISDN-PBX" Problem

This problem type covers the following erroneous conditions:

◊ Hand over from cell to cell is not working
◊ Bad speech quality

Note	If the DECT Multi-Cell system is connected to an IP-PBX then DECT problems must be solved with the responsible of the IP-PBX.

DECT-Multi-Cell systems connected to an ISDN-PBX which is working with in a VoIP environment experience special issues. Most issues are interconnected with accuracy of the synchronization clock of the ISDN-PBX. If this synchronization clock is not especially precise then the reference clock of the DECT-Multi-Cell system will have problems as described above.

1 Step: Check the ISDN reference clock

	Customer	Internet ISP	Telephony Provider
Telephony			
Data Transfer			
Equipment	Checks: ◊ Does the ISDN-PBX take its clock reference from a high precision clock? Actions: Make sure the ISDN-PBX takes its reference clock from a high precision source. Use an ISDN-Gateway which provides a high precision clock.		

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Manual of the Aarenet VoIP Switch Support Tools

VoIP Switch ConfigCenter Support Tools

The ConfigCenter Support Log

The "Support Log" provides the supporter with information from the internal processes of the ServiceCenter:

◊ Registration
◊ Connection setup, release and exceptions
◊ Call Routing
◊ Used Ruleset
◊ Emergency calls
◊ etc

The "Support Log" provides filters for:

◊ Time based selection: From ? Until, From ? Duration
◊ Text filter
◊ Registration events
◊ Call events
◊ etc.

The "Support Log" has a limited history. The history may last from a few hours up to some days. The length of the history may be different from VoIP switch to VoIP switch and depends on the length of log files and amount of logging events.

Note	The "Support Log" is tenant sensitive. This means a supporter of tenant A is not able to see events of tenant B!

Navigate to the "Support Log"

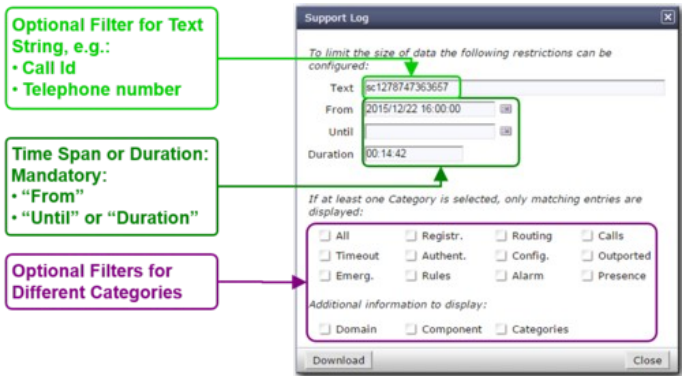
ConfigCenter:

→ Menu "Support"

→ Menu "Support Log"

Get a "Support Log"

Dialog: "Support Log":



When the dialog "Support Log" opens it contains by default in "From" the actual date/time (-5min) and in "Duration" a duration of 5min:

1. Click the Button [Download]
2. Via HTTP an ASCII formatted file with the last 5 minutes will be downloaded

Retrieving a "Support Log" in the past:

1. Insert the in "From" the desired start date/time
2. Insert in "Duration" the needed length
3. Press on the PC keyboard the 'Enter' key : The "Until" date/time will be computed
4. Click the Button [Download]

or

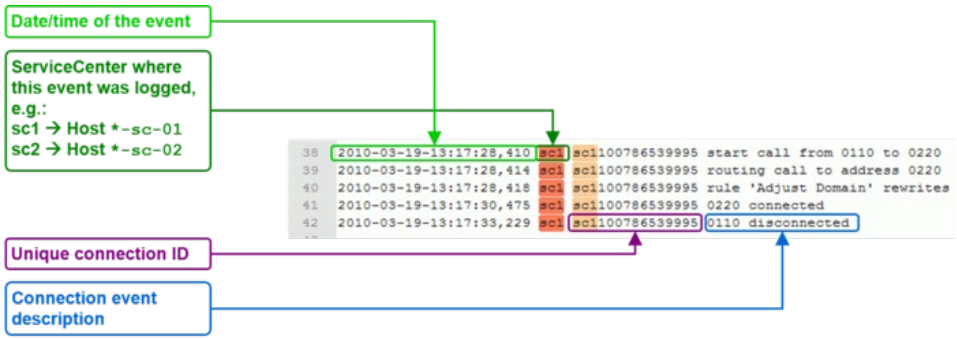
1. Insert the in "From" the desired start date/time
2. Insert the in "Until" the desired stop date/time
3. Press on the PC keyboard the 'Enter' key: The "Duration" will be computed
4. Click the Button [Download]

Best Practice	Get the events of a connection in the past: 1. Search the Call ID of the connection in the "Call Data" 2. Use the Call ID in the "Text" filter of the Support Log dialog 3. Make sure that the connection date/time match "From"- "Until" 4. Download the Support Log Get the events of a just finished connection: 1. Set the "Duration" to 5min (or shorter) 2. Download the Support Log 3. Search for the connection

Interpretation of a "Support Log"

The interpretation of a "Support Log" is quite easy and straight forward. With a little experience one will be soon familiar with the interpretation.

Interpretation and example of a call setup and release:



ConfigCenter Trace

The "Trace" provides the supporter with information from the message traffic between the VoIP switch and external VoIP devices, such as PSTN gateway, SIP CPE, SIP or MGCP telephones.

The "Trace" contains:

◊ Session Initiation Protocol SIP registration and connection signaling messages
◊ Media Gateway Control Protocol MGCP audit and endpoint control messages
◊ Session Description Protocol SDP streaming media initialization parameters

The "Trace" provides filters for:

◊ Time based selection: From ? Until, From ? Duration
◊ Text filter

The "Trace" has a limited history. The history may last from a few hours up to some days. The length of the history may be different from VoIP switch to VoIP switch and depends on the length of log files and amount of logging events.

The interpretation of a "Trace" (PCAP formatted file) has to be done in an external application like Wireshark network protocol analyzer. Wireshark offers deep and rich VoIP analysis .

Note	The "Trace" is **not** tenant sensitive. This means a supporter of tenant A is able to see signaling messages of tenant B! Due to this open display of information it may be possible that the "Trace" is not available for the supporters and operators on a multi tenant VoIP Switch.

Navigate to the "Trace"

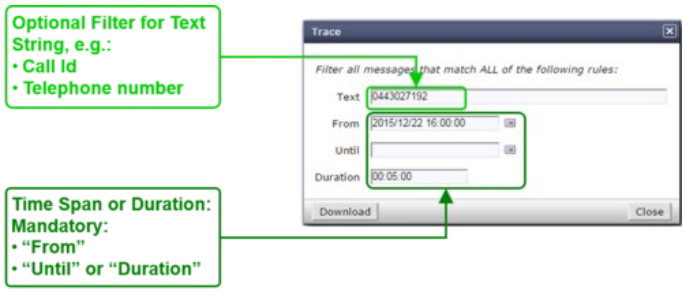
ConfigCenter:

→ Menu "Support"

→ Menu "Trace"

Get a "Trace"

Dialog: "Trace":



When the dialog "Trace" opens it contains by default in "From" the actual date/time (-5min) and in "Duration" a duration of 5min:

1. Click the Button [Download]
2. Via HTTP an PCAP formatted file with the last 5 minutes will be downloaded

Retrieving a "Trace" in the past:

1. Insert the in "From" the desired start date/time
2. Insert in "Duration" the needed length
3. Press on the PC keyboard the 'Enter' key: The "Until" date/time will be computed
4. Click the Button [Download]

or

1. Insert the in "From" the desired start date/time
2. Insert the in "Until" the desired stop date/time
3. Press on the PC keyboard the 'Enter' key: The "Duration" will be computed
4. Click the Button [Download]

Best Practice	Get the events of a connection in the past: 1. Search the connection in the "Call Data" 2. Click the Button [Trace] Get the events of a just finished connection: 1. Set the "Duration" to 5min (or shorter) 2. Download the Trace 3. Search for the connection

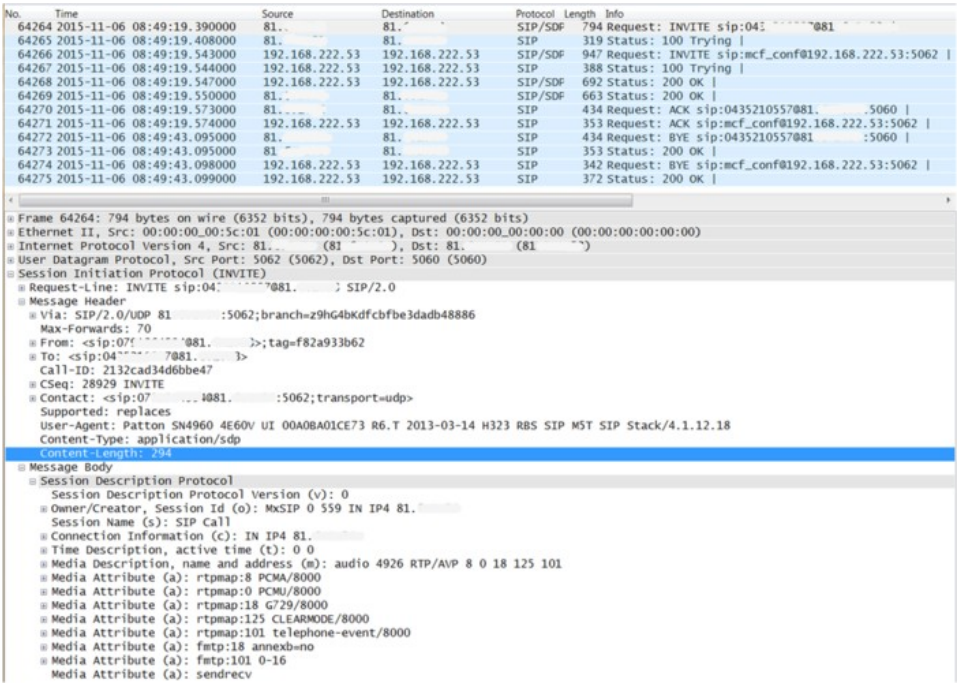
Interpretation of a "Trace"

The interpretation of a "Trace" needs experience!

For more information:

◊ See also article "Brief Tutorial of the SIP Signaling and SDP Media Protocols"
◊ Get a Wireshark training

Example of a Wireshark call capture, SIP setup and release:

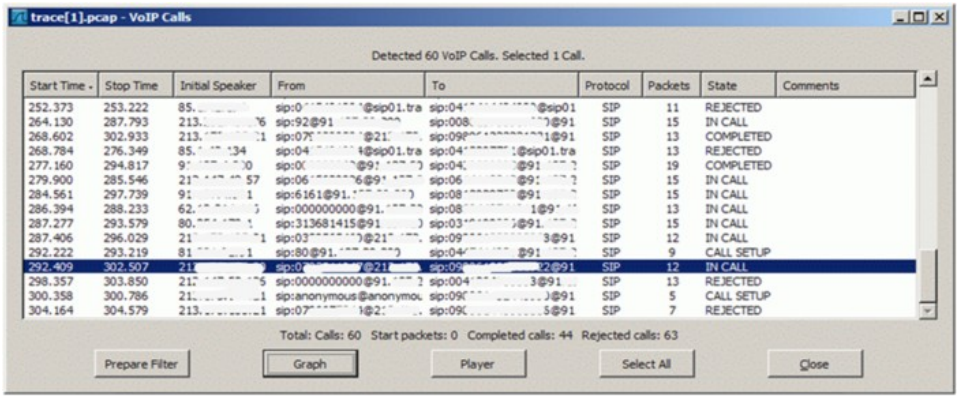


Example of a Wireshark call list:

Navigate in Wireshark:

- → Menu "Statistics"
  - → Menu "VoIP Calls"

Wireshark dialog where all calls are listed of the actual trace:

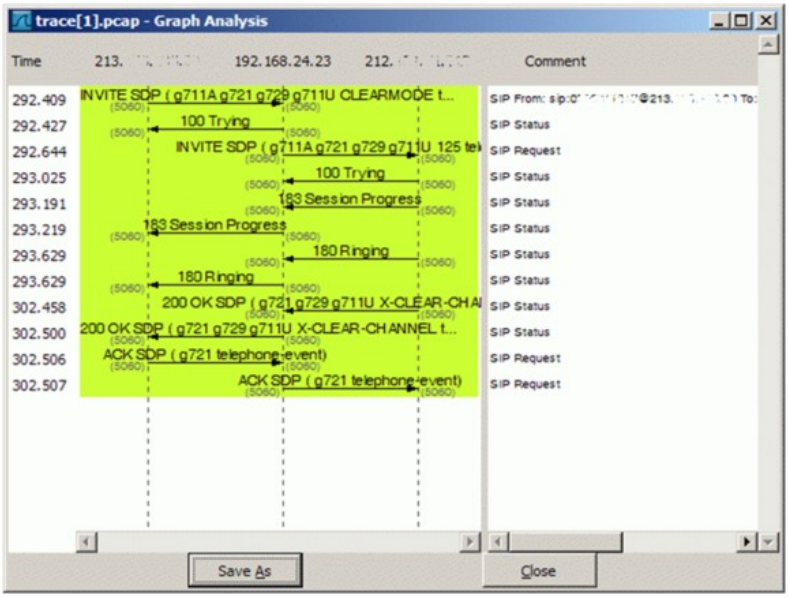


Example of a Wireshark call flow:

Navigate in Wireshark:

- → Menu "Statistics"
  - → Menu "VoIP Calls"
    - → Select the call of interest
      - → Click Button [Graph]

Wireshark dialog where the message flow is shown of the selected call:



The ConfigCenter Call Data

The "Call Data" lists the CDR of all incoming or outgoing connections or connection attempts. Extended filters enable the supporter to search for specific calls. The filters can be combined with logical AND.

Filter CDRs according:

◊ Call start and end date/time
◊ Call duration
◊ Call charges
◊ Telephone number of caller and/or callee.
◊ Tenants & account
◊ Price list attributes "Destination Type" & "Destination"

The "Call Data" has a limited history. The length of the history may be different from VoIP switch to VoIP switch and depends on the CDR storage length in the date base.

Selected CDR details allow direct access to the information of:

◊ SIP Trace:
The SIP message contents of this specific connection or call attempt is shown. For the interpretation of the trace consult the article "Brief Tutorial of the SIP Signaling and SDP Media Protocols", chapter "Knowhow SIP Signaling" .
◊ RTP/RTCP Media:
The RTP/RTCP information and statistics of this specific connection or call attempt is shown. For the interpretation of the media information consult the article "Brief Tutorial of the SIP Signaling and SDP Media Protocols", chapter "Knowhow Media Stream" .

Note	• The "Call Data" has a limited history. The length of the history may be different from VoIP switch to VoIP switch and depends on the CDR storage length in the date base. • Not all filter options may be available on the VoIP Switch. • The "Call Data" is tenant sensitive. This means a supporter/operator of tenant A is not able to see events of tenant B!

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Navigate to the "Call Data"

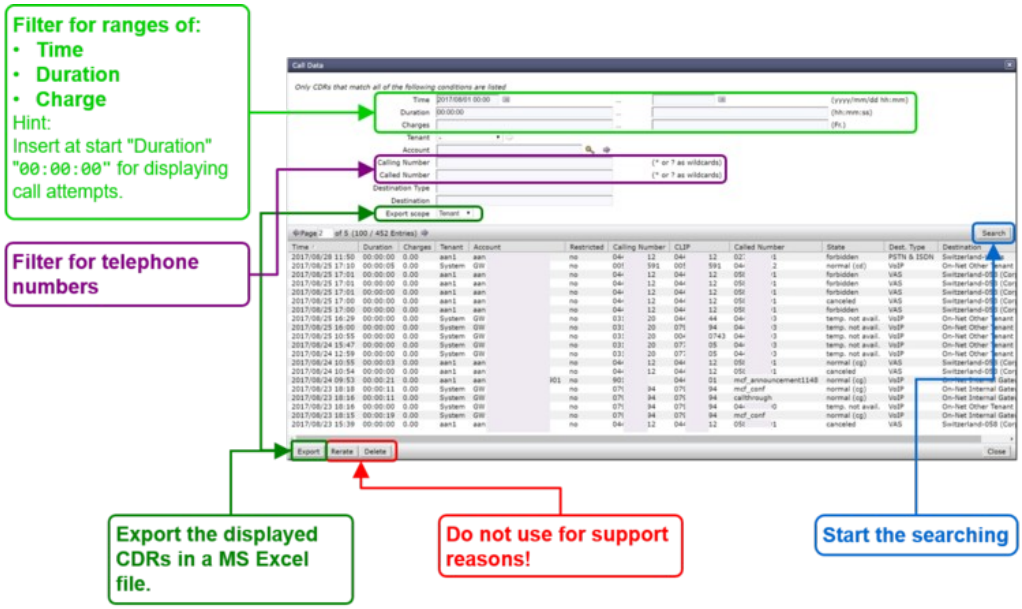
ConfigCenter:

→ Menu "Rating"

→ Menu "Call Data"

Get the "Call Data"

Dialog: "Call Data":



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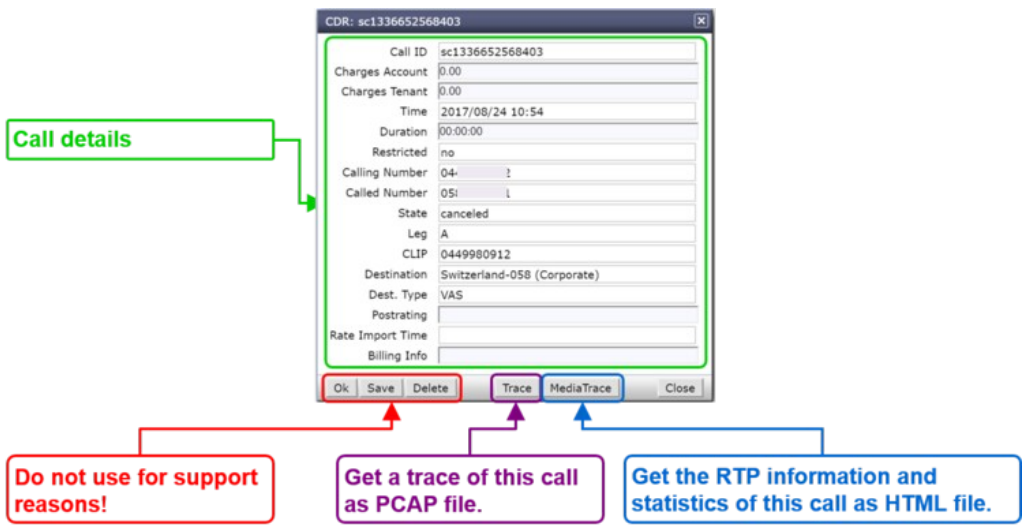

-

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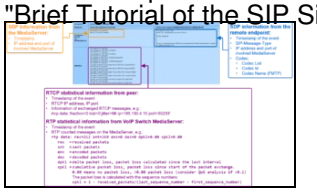
- 

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The ConfigCenter Address Registration

The ConfigCenter "Address Registration" displays if a SIP device or MGCP MTA has registered the telephone number. The supporter finds the following information of the registering devices:

◊ Type of registration, SIP, notifications, presence, etc
◊ IP address
◊ SIP user agent
◊ Registration time left.

Registrations can be de-registrated on the VoIP Switch by force.

Hint:
The device cannot be informed that it was de-registerd on the VoIP Switch. That means you have to wait until it re-registers automatically or force the device manually to re-register.

Navigate to "Registrations"

ConfigCenter:

→ Menu "Addresses"

or

→ Menu "Accounts"

→ Click on the line of the desired account

→ Click on the right arrow at "Addresses"

For details:

→ Click on the line of the desired address

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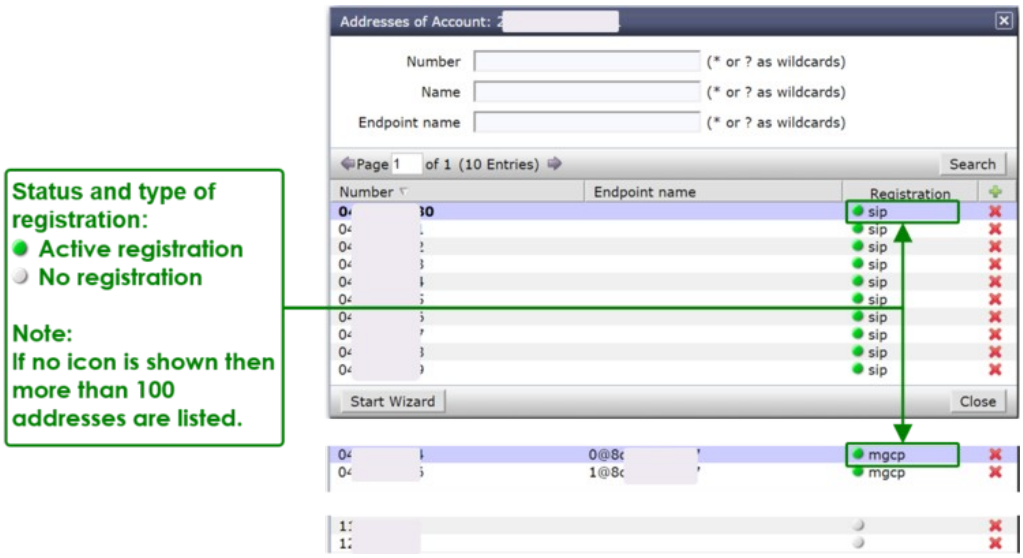
Interpretation of "Registrations" Information

Display of "Addresses" and registration overview:



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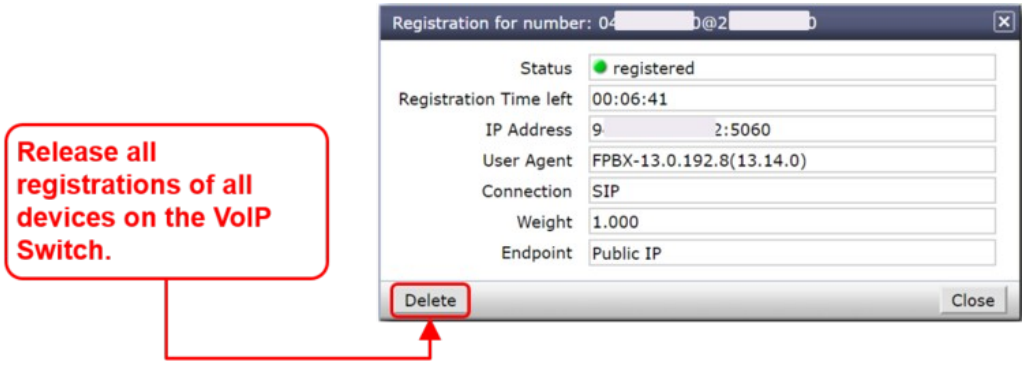


The ConfigCenter Components

The "Components" displays the state and activity of the VoIP Switch components. The components are the entities of the VoIP Switch that provide all functionality and features. The display is automatically updated every few seconds and shows the actual state and load of every component.

Note	On most VoIP Switches the "Components" display is not available for the supporters and operators.

Navigate to "Components"

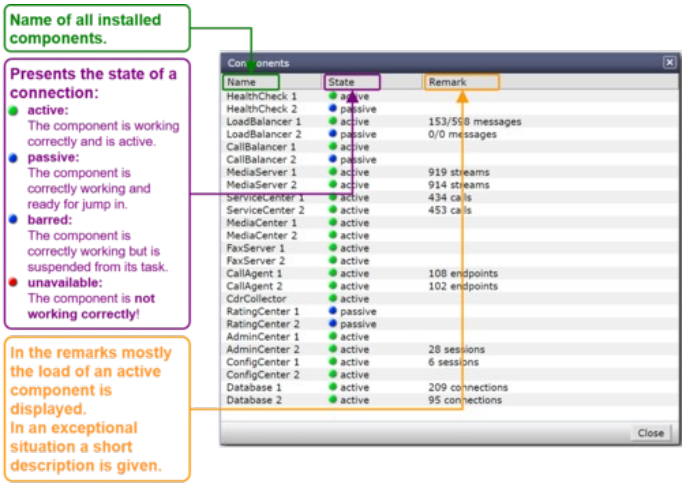
ConfigCenter:

→ Menu "System"

→ Menu "Components"

Interpretation of "Components" Information

Display of "Components":



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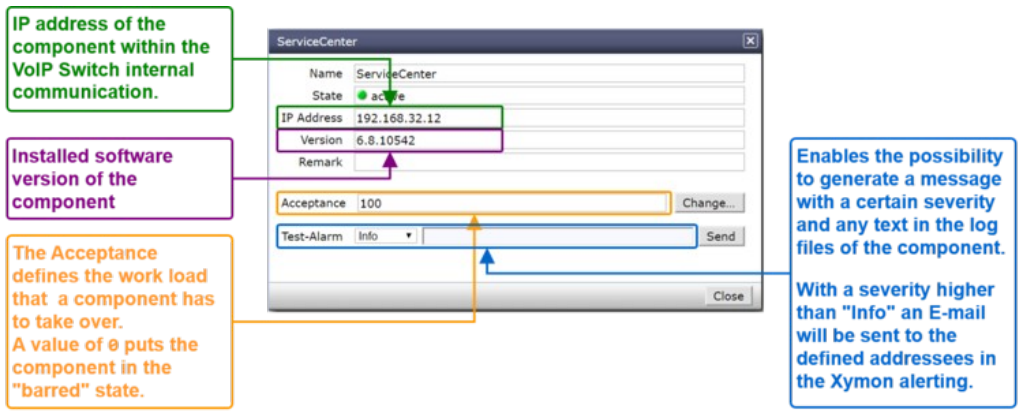

The ConfigCenter Channels

The ConfigCenter "Channels" is a live display of the current active connections and connection build-up. The administrator can filter an search the connections. If needed a connection can be forced to be released.

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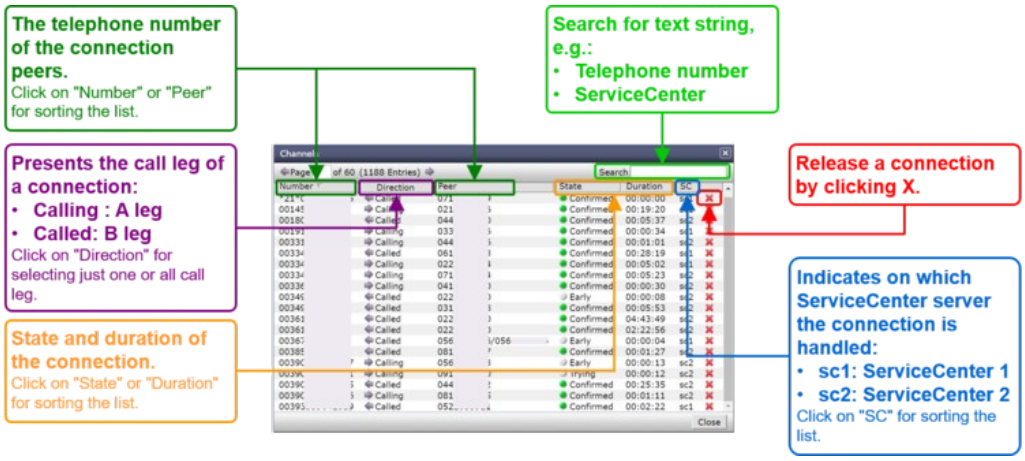
Navigate to "Channels"

ConfigCenter:

→ Menu "Channels"

Interpretation of "Channels" Information

Display of "Channels":



The ConfigCenter System Utilization

The "System Utilization" gives a statistical overview of the VoIP Switch resource utilization:

◊ Number of accounts
◊ Number of addresses (telephone numbers)
◊ Number of registrations
◊ etc

Note	On most VoIP Switches the "System Utilization" display is not available for the supporters and operators.

Navigate to "System Utilization"

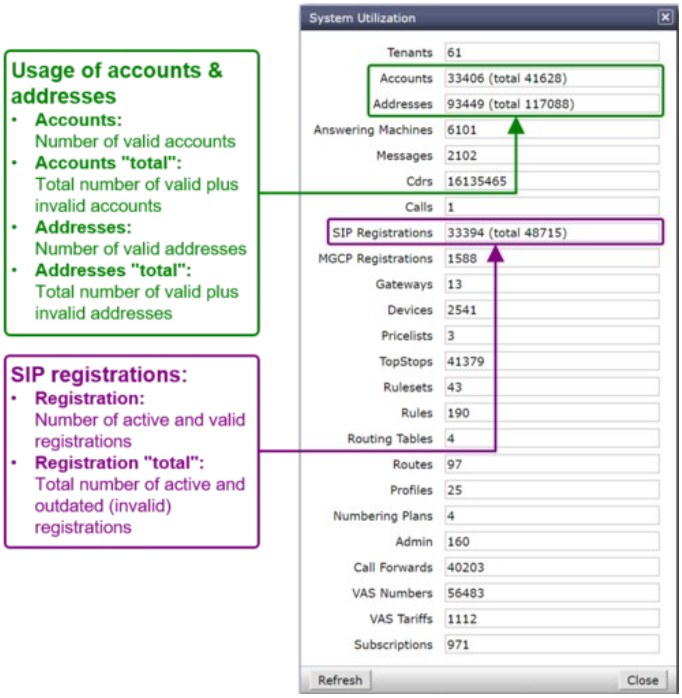
ConfigCenter:

→ Menu "System"

→ Menu "Utilization"

Interpretation of the "System Utilization" Information

The "System Utilization" provides the numbers of used resources:




```
'Enter' key: The "Duration" will be computed
Click the Button
```

Best Practice	Get the events of a connection in the past: 1. Search the connection in the "Call Data" 2. Click the Button [Trace] Get the events of a just finished connection: 1. Set the "Duration" to 5min (or shorter) 2. Download the Trace 3. Search for the connection

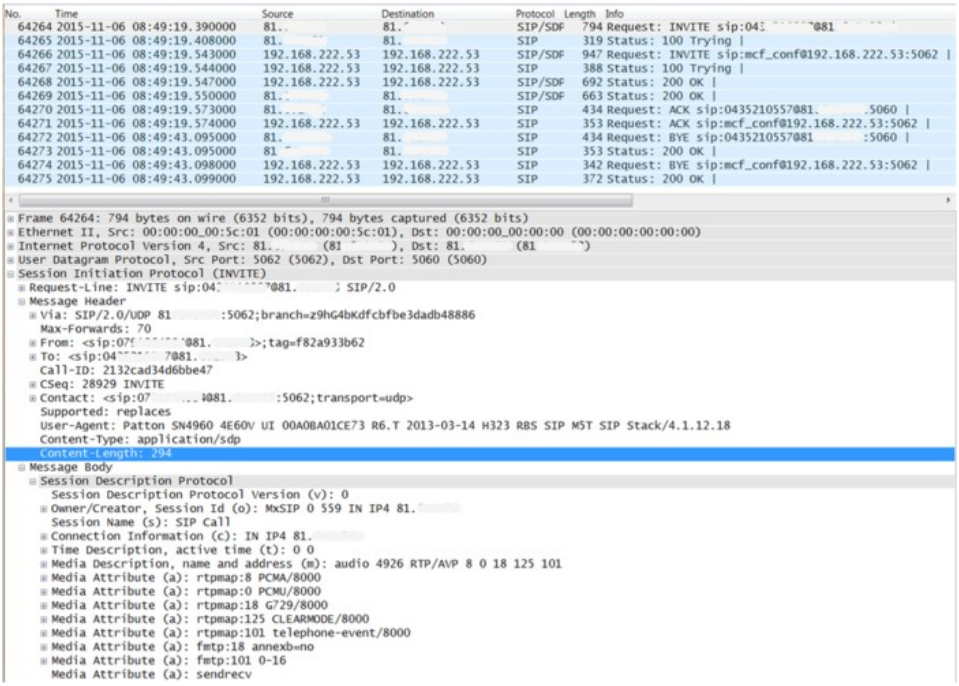
Interpretation of a "Trace"

The interpretation of a "Trace" needs experience!

For more information:

◊ See also article "Brief Tutorial of the SIP Signaling and SDP Media Protocols"
◊ Get a Wireshark training

Example of a Wireshark call capture, SIP setup and release:

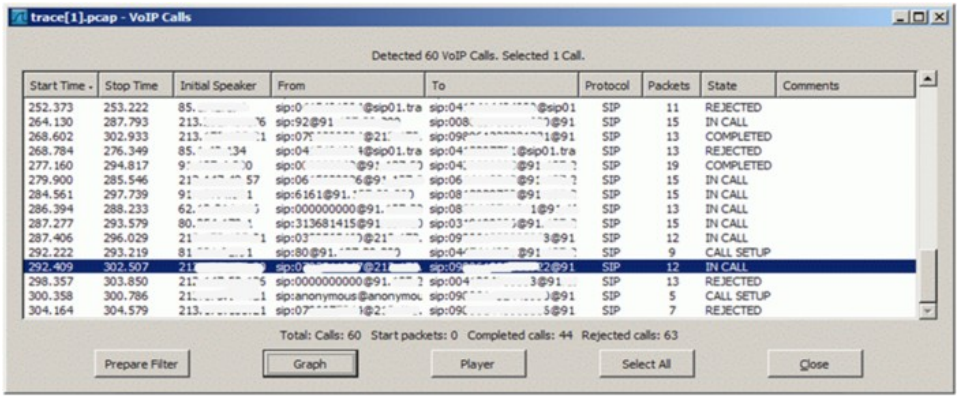


Example of a Wireshark call list:

Navigate in Wireshark:

- → Menu "Statistics"
  - → Menu "VoIP Calls"

Wireshark dialog where all calls are listed of the actual trace:

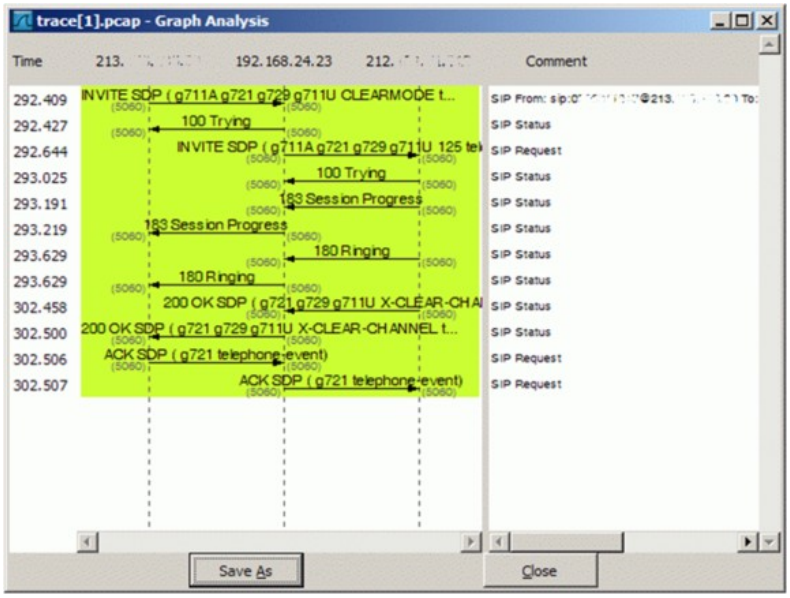


Example of a Wireshark call flow:

Navigate in Wireshark:

- → Menu "Statistics"
  - → Menu "VoIP Calls"
    - → Select the call of interest
      - → Click Button [Graph]

Wireshark dialog where the message flow is shown of the selected call:



The ConfigCenter Call Data

The "Call Data" lists the CDR of all incoming or outgoing connections or connection attempts. Extended filters enable the supporter to search for specific calls. The filters can be combined with logical AND.

Filter CDRs according:

◊ Call start and end date/time
◊ Call duration
◊ Call charges
◊ Telephone number of caller and/or callee.
◊ Tenants & account
◊ Price list attributes "Destination Type" & "Destination"

The "Call Data" has a limited history. The length of the history may be different from VoIP switch to VoIP switch and depends on the CDR storage length in the date base.

Selected CDR details allow direct access to the information of:

◊ SIP Trace:
The SIP message contents of this specific connection or call attempt is shown. For the interpretation of the trace consult the article "Brief Tutorial of the SIP Signaling and SDP Media Protocols", chapter "Knowhow SIP Signaling" .
◊ RTP/RTCP Media:
The RTP/RTCP information and statistics of this specific connection or call attempt is shown. For the interpretation of the media information consult the article "Brief Tutorial of the SIP Signaling and SDP Media Protocols", chapter "Knowhow Media Stream" .

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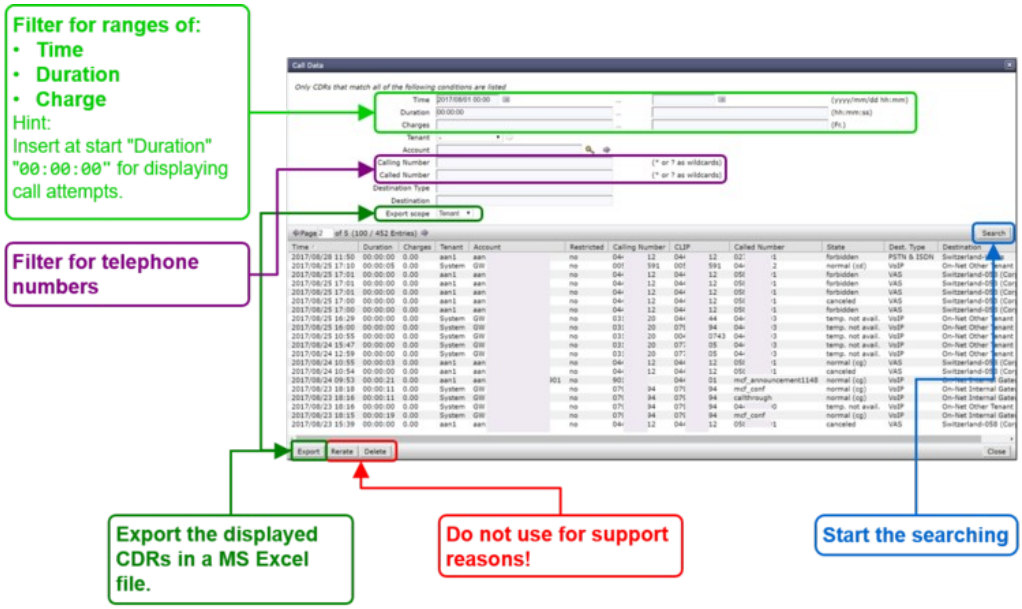

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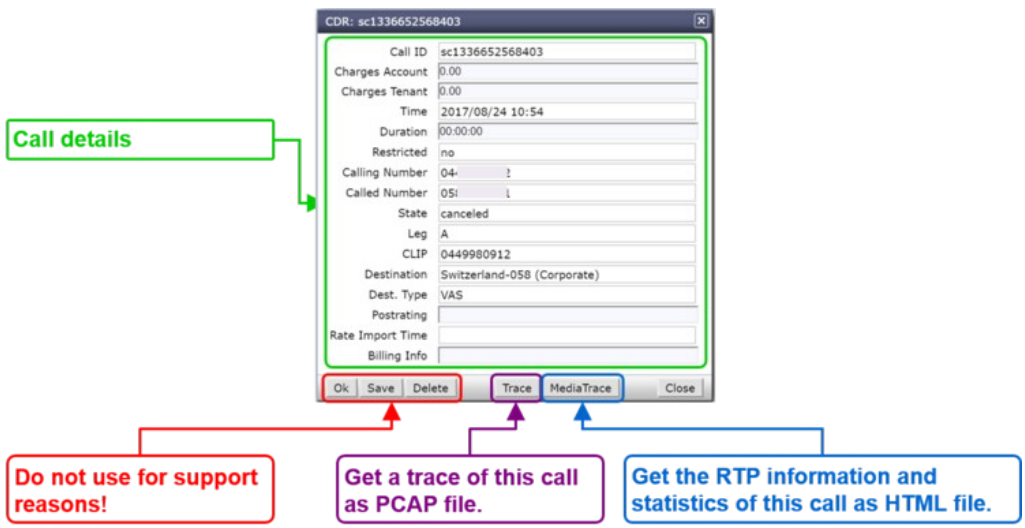

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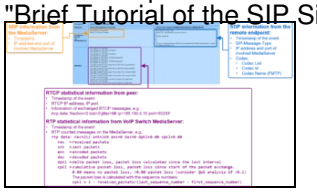
- 

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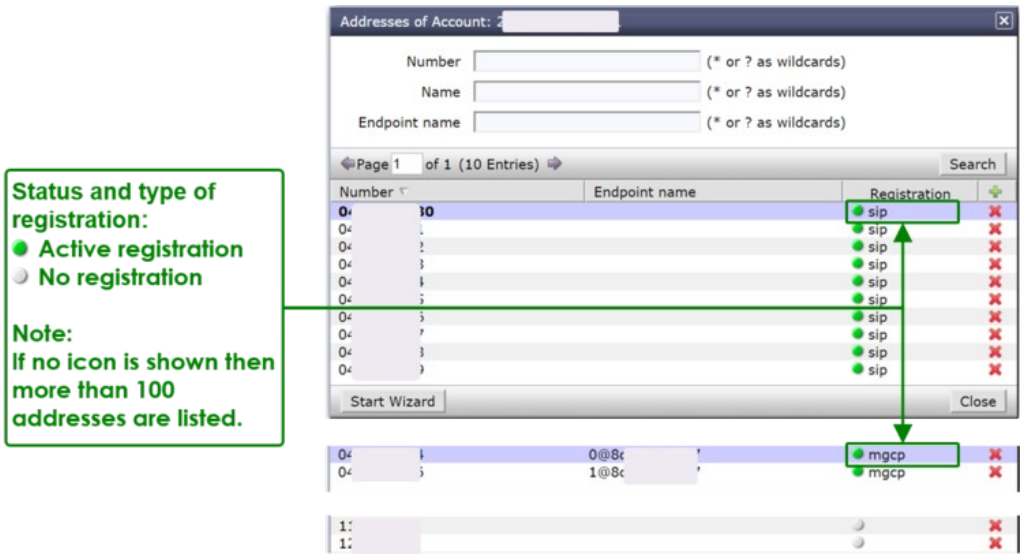

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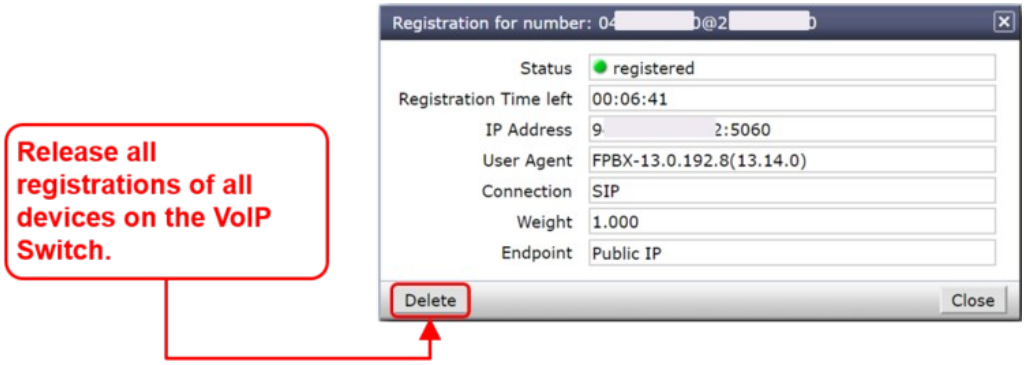

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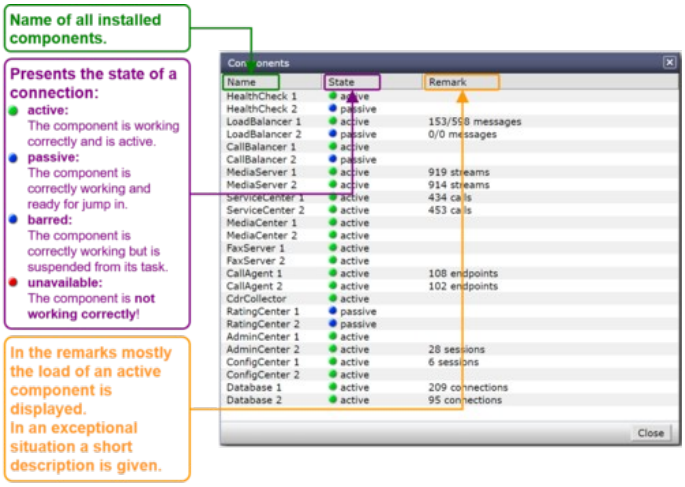

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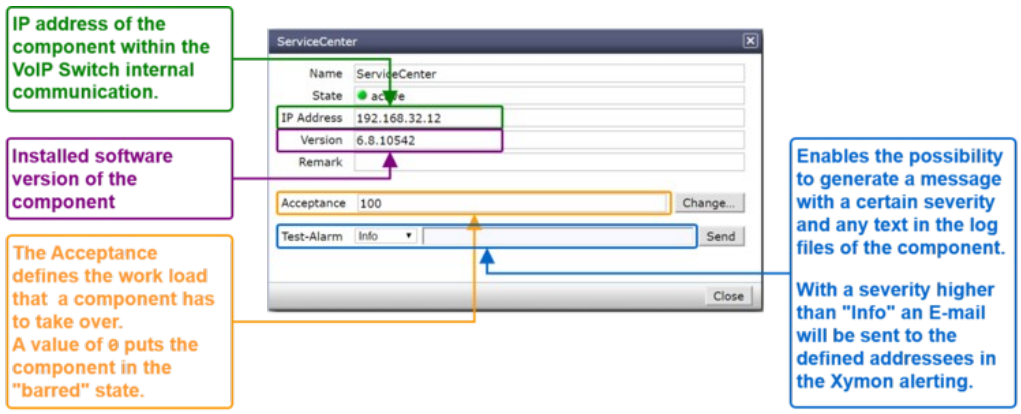

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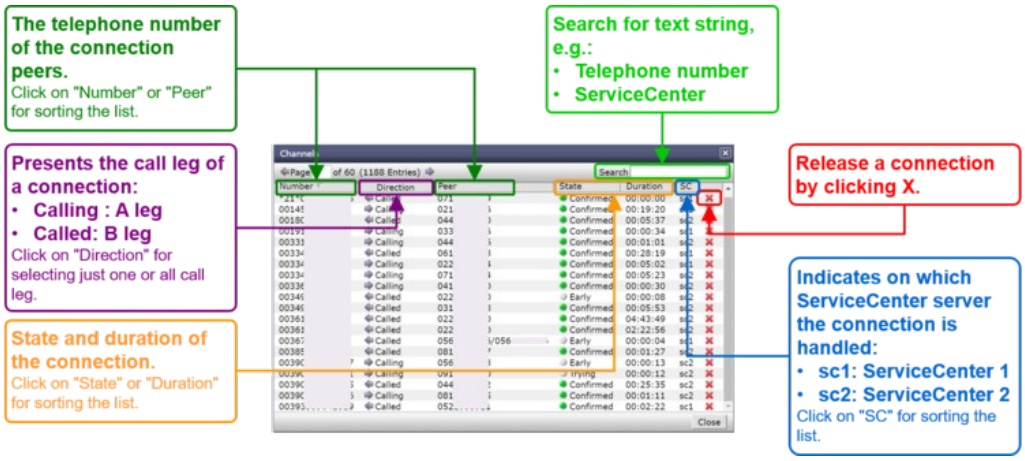

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Navigate to "System Utilization"

ConfigCenter:

→ Menu "System"

→ Menu "Utilization"

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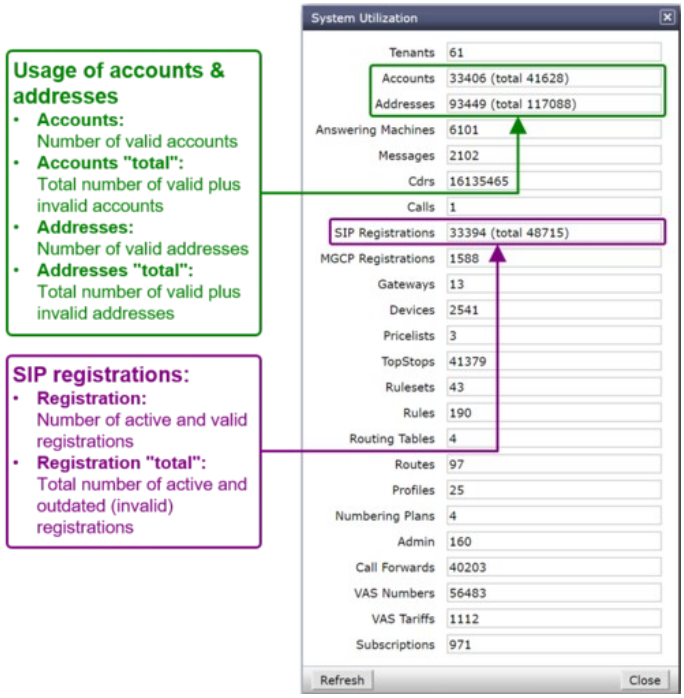

Usage of accounts & addresses
• Accounts:
Number of valid accounts
• Accounts "total":
Total number of valid plus invalid accounts
• Addresses:
Number of valid addresses
• Addresses "total":
Total number of valid plus invalid addresses
SIP registrations:
• Registration:
Number of active and valid registrations
• Registration "total":
Total number of active and outdated (invalid) registrations
System Utilization
Tenants 61
Accounts 33406 (total 41628)
Addresses 93449 (total 117088)
Answering Machines 6101
Messages 2102
Cdrs 16135465
Calls 1
SIP Registrations 33394 (total 48715)
MGCP Registrations 1588
Gateways 13
Devices 2541
Pricelists 3
TopStops 41379
Rulesets 43
Rules 190
Routing Tables 4
Routes 97
Profiles 25
Numbering Plans 4
Admin 160
Call Forwards 40203
VAS Numbers 56483
VAS Tariffs 1112
Subscriptions 971
Refresh
Close

-->

Manual for the Maintenance and Problem Solving of the Aarenet VoIP Switch

VoIP Switch Component Handling

Warning	All described actions can jeopardize the VoIP Switch's telephony service or server functionality! If there are uncertainties the contact the "VoIP Switch Supplier Support"

Basic VoIP Switch Component Commands

The VoIP Switch Administrator finds here instruction for VoIP Switch Component handling on OS console level:

◊ Start the VoIP Switch Component
◊ Stop the VoIP Switch Component
◊ Check the VoIP Switch Component status
◊ Restart the VoIP Switch Component
◊ etc

The VoIP Switch Component command affects only the instance on this server and can be executed with root rights only!

Command syntax:

```
root# <AS_COMPONENT> <COMMAND_OPTION>
```

Example:

```
root# configcenter status
```

Warning	Do not use other VoIP Switch Component command options as they can produce heavy problems!

Command	Command Option	Remark
<AS_COMPONENT> e.g.: `configcenter`		VoIP Switch Component command
	`version`	Lists the VoIP Switch Component version
	`status`	Lists the VoIP Switch Component status and process ID
	`stop`	Stops the VoIP Switch Component **The VoIP Switch Component stops immediately and any activity of the component will be interrupted!**
	`start`	Starts the VoIP Switch Component The VoIP Switch Component becomes immediately active and operative!
	`startpassive`	Starts the VoIP Switch Component but it remains passive.

		For becoming operative the VoIP Switch Component has to be started with the `start` option. Not all VoIP Switch Components offer this option.
	`restart`	Stops and starts the VoIP Switch Component The VoIP Switch Component becomes immediately active and operative!
	`restartpassive`	Stops and starts the VoIP Switch Component but it remains passive. For becoming operative the VoIP Switch Component has to be started with the `start` option. Not all VoIP Switch Component offer this option.
	`error`	Opens the error log file of the VoIP Switch Component
	`log`	Opens the actual log file of the VoIP Switch Component

Put Out of / Back to Service a VoIP Switch Component in an Operative VoIP Switch

The VoIP Switch Administrator finds here instruction for putting out or back of a VoIP Switch Component.

Put Out of Service a VoIP Switch Component

There are two ways to put out of service a VoIP Switch Component:

Variant 1: "Stop it hard"

Action:

A) Stop and check the component via the shell:

```
root# <AS_COMPONENT> stop
root# <AS_COMPONENT> status
```

The consequences are that the component stops immediately its operative work and all its running tasks.

The following VoIP Switch components may be stopped this way without jeopardizing the telephony service:

◊ ConfigCenter
◊ AdminCenter
◊ DataAccessCenter
◊ MediaCenter
◊ RatingCenter
◊ DataBase

Note	Make sure that: • The second component is active • The VoIP Switch administrators, operators and supporters are informed which ConfigCenter, AdminCenter are active • The users are able to use the active AdminCenter • The provider's CRM is able to use the active DataAccessCenter • The active RatingCenter is producing the CDR

Variant 2: "Stop it gracefully"

Action:

A) Stop gracefully the component via the ConfigCenter.

For the following components do flip the "active ? passive" role:

- HealthCenter
- LoadBalancer
- CallBalancer

do:

```
ConfigCenter GUI   Menu "System"    Menu "Components"

         Click the active component HealthCheck

                Click the fat right arrow at "Make component passive"

                       Confirm by clicking Button [ Yes ]
```

For the following components do a "pre-bar":

- ◊ ServiceCenter
- ◊ MediaServer
- ◊ FaxServer
- ◊ CallAgent

do:

```
ConfigCenter GUI   Menu "System"    Menu "Components"

         Click the desired VoIP Switch component

                Change the parameter "Acceptance" to 0
```

C) Wait until the component displays no activity anymore.

```
ConfigCenter GUI   Menu "System"    Menu "Components"
```

D) Stop and check the component via the shell:

```
root# <AS_COMPONENT> stop
root# <AS_COMPONENT> status
```

Put Back to Service a VoIP Switch Component

There are two ways to put back to service a VoIP Switch Component:

Variant 1: "Start it"

Action:

A) Start and check the component via the shell:

```
root# <AS_COMPONENT> start
root# <AS_COMPONENT> status
```

The consequence is that the component starts immediately its operative work.

Variant 2: "Start it gracefully"

This variant may make sense when the following VoIP Switch components shall become active but not operative immediately:

- ◊ ServiceCenter
- ◊ MediaServer
- ◊ FaxServer
- ◊ CallAgent

Action:

A) Start "passive" the component via the ConfigCenter.

```
root# <AS_COMPONENT> startpassive
root# <AS_COMPONENT> status
```

B) Make the component operative at the appropriate time:

```
ConfigCenter GUI   Menu "System"    Menu "Components"

         Click the desired VoIP Switch component

               Change the parameter "Acceptance" to 100
```

The "Acceptance" may by any value >0 according. Choose according the load balancing scheme of the component.

C) Check if the component displays activity:

```
ConfigCenter GUI   Menu "System"    Menu "Components"
```

Work Flow for Analyzing VoIP Switch Problems

Note	Not every red alarm jeopardizes the telephony service as a whole but a bulk of yellow warnings may endanger it!

The VoIP Switch Administrator and other service personnel find here a work flow for analyzing VoIP Switch problem indications and find out the appropriate action.

The main task is to find out if:

1. The situation jeopardizes the telephony service as a whole, e.g.:
   - · IP network issues
   - · Several VoIP Switch servers failed or off line
2. The database replication is broken
   - · IP network issues
   - · Server with running database failed
   - · Linux service MySQL failed
3. The situation hampers the operation of configuration of customer accounts, addresses etc.
   - · Management server failed or off line
   - · VoIP Switch component ConfigCenter, AdminCenter DataAccessCenter, RatingCenter stopped working correctly

The VoIP Switch Administrator finds here the work flow for analyzing VoIP Switch problems:

Work Flow for Analyzing VoIP Switch Problems

Analysis:

1. Check if it is a **single alarm** or a **bulk alarm** situation.

   a) Connect to the VoIP Switch monitor Xymon "Main View"

As a rule of thumb: It is a single error if only one issue is displayed.

2. Analyze and treat a **single alarm** situation:

a) Check the contents of the error message.

b) Compare the error description against the **Indication "Xymon Event"** ones in chapter "VoIP Switch Maintenance"

c) Check if the actual situation is equal or similar as described and the recommended actions suitable.

d) Execute the suitable actions.

If you are not sure contact the "VoIP Switch Supplier Support"

3. Analyze the **bulk alarm** situation:

a) Get a first overview of the situation by analyzing the Xymon Monitor :

Check in the MS-01 Xymon monitor the server, component and IP status:

```
Xymon GUI    Xymon "Main View"
```

1. Which type of server are affected?
   - At least one LoadBalancer LB server must be active that the telephony service can work!
   - At least one ServiceCenter SC server must be active that the telephony service can work!
   - At least one server with the operative database must be active that the telephony service can work!

2. Check the CPE registration statistic :
   - Do drop the CPE registrations?

3. Check the call statistic:
   - Do drop the VoIP Switch number of calls?

     ```
     Xymon GUI   Management Server   Column "calls_sys"
     ```
   - Do drop the calls on one or more ServiceCenter?

     ```
     Xymon GUI   ServiceCenter Server   Column "calls_sc"
     ```
   - Do drop the calls on one or more gateways?

     ```
     Xymon GUI   Gateway   Column "calls_gw"
     ```

4. Do the same check as above on MS-02 Xymon Monitor

5. Does the comparison of the two Xymon Monitor point out that:
   - The same single component on the same server failed?
   - All components of one side failed?
   - The Xymon Monitor sees only the components on its side?
   - The telephony service is running at least on one side

b) Extend the overview by analyzing the ConfigCenter "System Component" Overview :

Check in the MS-01 ConfigCenter the status of the VoIP Switch components:

```
ConfigCenter GUI   Menu "System"    Menu "Components"
```

1. Are actually calls running and new calls can be established?

2. Make test calls:
   - To and from a telephone number in the PSTN
   - On-net test calls
   - Call a well known VoiceMail Box from on-net and from PSTN

3. Is the number of running calls fast dropping and no new calls are established?

4. Which type of VoIP Switch components are affected?
   - At least one LoadBalancer component must be active that the telephony service can work!
   - At least one ServiceCenter component must be active that the telephony service can work!
   - At least one operative database must be active that the telephony service can work!

- Does this picture correspond to the results of the first overview in the Xymon Monitor ?

5. Do the same check as above on MS-02 ConfigCenter

6. Does the comparison of the two ConfigCenter point out that:
   - The same single component on the same server failed?
   - All components of one side failed?
   - The ConfigCenter sees only the components on its side?
   - The telephony service is running at least on one side

4) Treat **bulk alarm** situations:

a) Is there a VoIP Switch server hardware, RAID or hard-disk problem?

Indications:

Indication:
`<HOST_NAME> "snmptrapd" "failure"`
`<HOST_NAME> "snmptrapd" "degraded"`

Actions:

For DELL server see: "Treating Problems of Servers from DELL Inc ®"

b) Is the IP connectivity affected to or between VoIP Switch servers?

Note	If VoIP Switch servers are affected then a lot of additional alarming messages of missing VoIP Switch components will pop up!! **This can be one of the most annoying erroneous situations!**

Indications:

Indication:
`<HOST_NAME> conn "Host does not respond to ping" <IP_ADDRESS>`
* Dropping CPE registrations ! • Dropping calls ! • No new calls!

Actions:

See: "Maintenance Due to IP Network Alarm"

c) If you are not sure what to do then contact the "VoIP Switch Supplier Support"

VoIP Switch Server Maintenance

Maintenance Due to VoIP Switch Components General Alarms

Maintenance Due to Messages from Java Framework

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> msgs "Jdbc"

Description:
Java internal exceptions. Mostly due to database accesses which are hopefully handled by the application.

Consequences:
For the VoIP Switch telephony service:

◊ Mostly none

For the operations:

◊ Mostly none

For the user:

◊ Mostly none

Solution:
Observe the frequency of this event

Action:
1. Observe the frequency of this event

2. If the erroneous condition is to frequent then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Messages from VoIP Switch Components Internals

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> msgs "EventQueue"
<HOST_NAME> msgs "SysCompDatabase - Cannot evalute status"

Description:
These events may happen on all VoIP Switch servers and are VoIP Switch component internal notes.

Consequences:
For the VoIP Switch telephony service:

◊ Mostly none

For the operations:

◊ Mostly none

For the user:

◊ Mostly none

Solution:
Observe the frequency of this event

Action:
1. Observe the frequency of this event

2. If the erroneous condition is to frequent then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Messages from LoadBalancer Server

Maintenance Due to HealthCheck Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> msgs "HealthCheck"`

Description:
The HealthCheck supervises the status of virtual IP addresses and their associated physical IP addresses. If the HealthCheck on one server doesn't see the peer physical IP address it takes over the virtual IP address. It most probably points out an IP network problem in the "Public Voice Segment"

Consequences:

Warning	This erroneous condition must be checked within reasonable time!

For the VoIP Switch telephony service:

- ◊ None if concurrently no other IP network problems arise

For the operations:

- ◊ None

For the user:

- ◊ None

Solution:
Solve the IP network if needed.

Check status the VoIP Switch component with an active-passive scheme:

- ◊ LoadBalancer
- ◊ CallBalancer
- ◊ RatingCenter

Action:
1. Check if the IP network is OK

2. Check the status of the LoadBalancer components

   Confirm if the active LoadBalancer swapped, e.g. from *-lb-01 to *-lb-02

3. Check the status of the CallBalancer components

Confirm if the active CallBalancer swapped, e.g. from *-lb-01 to *-lb-02

4. Check the status of the RatingCenter components

Confirm if the active CallBalancer swapped, e.g. from *-ms-01 to *-ms-02
Confirm if the active RatingCenter is processing the CDR's

5. Treat the problem:

a) If there are IP network problems

Actions see: "Maintenance Due to IP Network Alarm"

b) If there is a LoadBalancer problem try to restart the component:

```
root# loadbalancer restart
```

c) If there is a CallBalancer problem try to restart the component:

```
root# callbalancer restart
```

d) If there is a RatingCenter problem try to restart the component:

```
root# ratingcenter restart
```

e) If the RatingCenter swapped make sure that the CDR are processed:

1. ConfigCenter GUI Menu "System" Menu "Components"
Click line at "active" RatingCenter -> In dialog select "Process CDRs"
Click button [Close]
2. The CDR CSV-Files are processed:

```
root# cd /home/servicecenter/cdrs
```

Check if the CSV files have an actual time stamp which indicates that new CDRs where written:

```
root# ls -ltra
```

Open a CSV file and check for new entries, e.g.:

```
root# less monthly.csv
```

6. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

If those events are logged subsequently then rapport it to the "VoIP Switch Supplier Support"!

Maintenance Due to LoadBalancer Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> msgs "Balancer"
```

Description:

LoadBalancer internal problem that is treated internally by the component. The LoadBalancer has an "active-passive" redundancy scheme.

Consequences:
For the VoIP Switch telephony service:

◊ None

For the operations:

◊ None

For the user:

◊ None

Solution:
Not defined yet

Action:
1. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

If those events are logged subsequently then rapport it to the "VoIP Switch Supplier Support"!

Maintenance Due to LoadBalancer Message "Missing ServiceCenter"

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> msgs "BalancerSwitch" <SERVICECENTER> "not
available anymore"
```

Description:
The LoadBalancer indicates that it doesn't see a certain ServiceCenter.

This happens when:

◊ the ServiceCenter has restarted
the event will be transient
◊ the ServiceCenter is stopped
the event will remain until the ServiceCenter is started again
◊ no IP connectivity
the event will remain until the IP connectivity is reestablished

Consequences:

Warning	This erroneous condition must be handled within reasonable time!

For the VoIP Switch telephony service:

◊ None, the other ServiceCenter take over the work load
◊ If a ServiceCenter is missing then the VoIP Switch looses redundancy capability

For the operations:

◊ None

For the user:

◊ None

Solution:
Solve the IP network problems if needed:

Actions see: "Maintenance Due to IP Network Alarm"

Solve the server problem if needed

Actions see: "Treating Server Hardware Problems"

Action:
1. Check if the IP network is OK

2. Check the status of the ServiceCenter components

Confirm that the reported ServiceCenter server is affected

3. Check the reported ServiceCenter server with the "Server Administrator (OMSA)"

4. Treat the problem:

a) If there are IP network problems

Actions see: "Maintenance Due to IP Network Alarm"

b) If there is a ServiceCenter problem try to restart the component:

```
root# servicecenter restart
```

5. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to CallBalancer Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> msgs

Description:
The CallAgent dispatches MGCP messages to the CallAgent components.

The CallAgent has an "active-passive" redundancy scheme.

Consequences:

Warning	This erroneous condition must be checked within short time!

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ None

For the user:

◊ Users with MGCP MTA as telephone adapter may not be able to telephone

Solution:
Check status the CallBalancer active-passive scheme and if the MGCP messages are processed.

Action:
1. Check if the IP network is OK

2. Check the status of the CallBalancer components:

a) Confirm if the active CallBalancer swapped , e.g. from *-ms-01 to *-ms-02

b) Confirm if the active CallBalancer is processing the MGCP messages

Check if the CallAgent treat MGCP connections and that the total number of MGCP connections is not dropping.

3. Check if the MGCP audits are not dropping:

a) Connect to a Xymon monitor and check in Xymon Column "regs" the numbers of MGCP-Active and MGCP-Brocken

b) Check the questions:

◊ Do drop the number of MGCP-Active?
If yes => There may be a IP backbone problem or CallBalancer, CallAgent outage!

4. Treat the problem:

a) If there are IP network problems

Actions see: "Maintenance Due to IP Network Alarm"

b) If there is a CallBalancer problem try to restart the component:

```
root# callbalancer restart
```

5. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to MediaServer Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> msgs "MediaConnection (06) Cannot handle outgoing message"`
`<HOST_NAME> msgs "MediaServerProvider (MS) refreshing mediaserver mc1ms2 failed"`

Description:
The MediaServer records or plays back announcements and VoiceMail messages. Occasionally it may not correctly record a message and transfer it to the MediaCenter or play back an announcement or message.

The MediaServer can act as media proxy for active connections and transcode media streams.

Consequences:

Warning	If in this VoIP Switch the MediaServer acts as media proxy then the erroneous situation must be checked soon!

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ None

For the user:

◊ A VoiceMail Box message or announcement couldn't correctly record or played back.
◊ User may not hear the other side or vica versa.

Solution:
Depends on the situation.

Action:
1. If the erroneous condition remains or happens to often then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Messages from Management Server

Maintenance Due to AdminCenter Message "Missing FMC Application Server"

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> msgs "FmcRequest - Cannot post request"`
`<HOST_NAME> msgs "FmcProvider - could not provision pbx"`

Description:
The AdminCenter tried to configure the FMC application.

Consequences:

Warning	This erroneous condition is sporadic or must be handled within reasonable time!

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ A configuration on a FMC server failed

For the user:

◊ A user "an MC-Phone" is not working

Solution:
Check the state of the FMC servers and their IP connectivity toward the VoIP Switch servers.

Action:
1. Check if the IP network is OK

2. Check the status of the FMC server

3. Treat the problem:

a) If there are IP network problems

Actions see: "Maintenance Due to IP Network Alarm"

b) If there is a FMC server problem

Contact the "VoIP Switch Supplier Support"!

4. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to AdminCenter Message "Missing Redirection Server"

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> msgs "FmcProvider - could not provision user"
<USER_TELEPHONE_NUMBER>
```

Description:
The mobile app "an MC-Phone" couldn't get the information from the associated redirection server (by default a Comdasys server located in Europe) where its responsible configuration server is located. Therefore the users "an MC-Phone" couldn't obtain its configuration and will not work.

Consequences:

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ None

For the user:

◊ The mobile app "an MC-Phone" will not work

Solution:
Make sure to have good IP connectivity to the Internet

Action:
1. The user must find a reliable Internet connection and restart the app "an MC-Phone" until it gets its configuration

Maintenance Due to ConfigCenter Message "Wrong User Login"

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> msgs "msgsAccessLogger - ADMIN:login; user"
<USERNAME> "-> User Blocked"
```

Description:
A VoIP Switch Administrator, Operator, Supporter tried to login to the ConfigCenter with wrong credentials. The user will be blocked for several minutes.

Consequences:

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ The user will be blocked from the ConfigCenter for several minutes.

For the user:

◊ None

Solution:
Wait

Action:
1. Retry after a few minutes with the correct login credentials.

2. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to ConfigCenter Message "DB Replication Check"

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> msgs JdbcReplicationMonitor "Replication"
'<BROKEN_REPLICATION_DIRECTION>' "is broken!"
```

Description:
The database replication check was not successful. This can happen from time to time when the database has to process heavy load.

In most cases the database replication recovers automatically even after several hours of failed replication. If it is not recovering then this is a severe problem and must be treated.

Consequences:

Warning	If this erroneous condition remains then this is a SEVERE erroneous condition and must be treated within short time!

For the VoIP Switch telephony service:

◊ The database redundancy is endangered

For the operations:

◊ None

For the user:

◊ None

Solution:
Restore the MySQl DB replication if the erroneous condition remains.

Action:
1. Check periodically (ca. every half hour) the Xymon monitor for this error condition.

2. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to DataAccessCenter Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> msgs "Jdbc" "SQL-Exception during statement"`

Description:
A configuration via the DataAccessCenter may have failed.

This may happen if the database is under heavy load.

Consequences:

Warning	This erroneous condition must be checked within reasonable time!

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ A customer configuration may have failed (which is hopefully covered by the CRM application).

For the user:

◊ None

Solution:
Inter-working between the DataAccessCenter and database must be optimized.

Action:
1. If this Java event is logged subsequently then rapport it to the "VoIP Switch Supplier Support"!

Maintenance Due to RatingCenter Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> msgs
```

Description:
The RatingCenter has an "active-passive" scheme. Every RatingCenter event has to be checked if the active RatingCenter is working correctly and is processing the CDRs.

Consequences:

Warning	This erroneous condition must be checked within short time!

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ A CDR may be not written correctly into the CDR database and/or CSV files.
◊ The customer billing contains not all CDR

For the user:

◊ None

Solution:
Check status the RatingCenter active-passive scheme and if the CDR are processed.

Action:
1. Check the status of the RatingCenter component

Confirm if the active RatingCenter is processing the CDR's

2. Treat the problem:

a) If the RatingCenter swapped make sure that the CDR are processed:

Open the ConfigCenter Menu "Components"

Click line at "active" RatingCenter -> In dialog select "Process CDRs"

Click button [Close]

b) Check if the CDR CSV-Files are processed:

Open the CDR directory:

```
root# cd /home/ratingcenter/cdrs
```

Check if the CSV files have an actual time stamp which indicates that new CDRs where written:

```
root# ls -ltra
```

Open a CSV file and check for new entries, e.g.:

```
root# less monthly.csv
```

3. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Messages from ServiceCenter Server

Maintenance Due to FaxServer Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> msgs

Description:
Fax may not received correctly. The mailing of the PDF file may fail.

Consequences:

For the VoIP Switch telephony service:

- ◊ None

For the operations:

- ◊ None

For the user:

- ◊ A received Fax may not be correctly received and transferred to the user. This situation is usually handled by the Fax device either automatically or manually.

Solution:
Restart the FaxServer component.

Action:
1. Check if no Fax at all are received.

   Send test fax.

2. Restart the FaxServer:

```
root# faxserver restart
```

3. If the FaxServer logs subsequently then rapport it to the "VoIP Switch Supplier Support"!

Maintenance Due to MediaCenter Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> msgs MediaCenterCall
<HOST_NAME> msgs MediaServer
<HOST_NAME> msgs "file not found"

Description:
The MediaCenter handles the WAV files from announcements and VoiceMail messages. Occasionally it may not correctly record a message, loose a message file. Also an order to the MediaServer may fail to replay a message

or announcement.

Consequences:

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ None

For the user:

◊ A VoiceMail Box message or announcement couldn't correctly recorded or played back

Solution:
Clean up the VioceMail message date base.

Optimize the inter-working of MediaCenter and MediaServer

Action:
1. If those events are logged subsequently then rapport it to the "VoIP Switch Supplier Support"!

Maintenance Due to ServiceCenter Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> msgs`

Description:
The ServiceCenter is the main component of the VoIP Switch. It computes the connections signaling and telephony features.

The ServiceCenter has an all active redundancy scheme. If one ServiceCenter fails the remaining ServiceCenter take over the work load.

Consequences:

Warning	This erroneous condition must be checked and treated within short time!

For the VoIP Switch telephony service:

◊ As long one ServiceCenter remains the VoIP Switch works!

For the operations:

◊ None

For the user:

◊ None

Solution:
Depends on the analyzed problem.

Action:
1. Check how acute the problem is:

a) Check if the IP network is OK

b) Check the status of the ServiceCenter component

◊ Are enough ServiceCenter active that the work load can be treated?
If NO then there is a most SEVERE erroneous situation

c) Check in the ConfigCenter Menu "Components" if the active ServiceCenter is processing the connections:

◊ Do drop the total number of connections?
If YES then there is a most SEVERE erroneous situation:
There may be a IP backbone problem!

d) Check in the Xymon Column "regs" the number of registered SIP-Devices:

◊ Do drop the number of SIP-Devices?
If YES then there is a most SEVERE erroneous situation:
There may be a IP backbone problem!

e) Check the reported ServiceCenter server with the "Server Administrator (OMSA)"

◊ Are problems signaled?

2. Treat the problem:

a) If there are IP network problems

Actions see: "Maintenance Due to IP Network Alarm"

b) If there is a ServiceCenter problem try to restart the component:

```
root# servicecenter restart
```

c) If there is a hardware problem:
Actions see: "Treating Server Hardware Problems"

3. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to ServiceCenter Message "License Violation"

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> msgs License "License Violation"
<HOST_NAME> msgs License "grace-period remaining:"
```

Description:
This ServiceCenter has a license problem and will work only for the remaining grace period.

Consequences:

Warning	This erroneous condition must be checked and treated within the remaining grace period!

For the VoIP Switch telephony service:

◊ As long one ServiceCenter remains the VOIP Switch works
◊ The telephony service will be stopped on this ServiceCenter after passing of the grace period

For the operations:

◊ None

For the user:

◊ None

Solution:
Get actual licenses from the VoIP Switch Supplier.

Action:
1. Check in the ConfigCenter Menu "Components" which ServiceCenter component has a license problem and how long the grace period is.

2. Contact the "VoIP Switch Supplier Support"!

Maintenance Due to ServiceCenter Message "Failed Emergency Call"

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> msgs ServicePrioCallControl "Could not establish
priority-call". Call from
Connection/<SIP_CALL_ID>/<CALLING_NUMBER> to
<CALLED_EMERGENCY_NUMBER>
```

Description:
A user's emergency call failed!

Consequences:

Warning	**Severe** legal condition that must be handled! This case can have legal consequences for the provider!

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ None

For the user:

◊ The emergency call did not work

Solution:
Check if the call routing failed due to a VoIP Switch emergency call treating or routing. If yes fix them.

Check if the PSTN provider did reject the emergency call. If yes contact the PSTN provider.

Action:
1. Archive traces for legal responsibilities:

◊ Save the trace of this emergency call and all subsequent calls from this user toward emergency numbers

2. Check where the call was rejected.

◊ If the call was rejected at the PSTN provider side contact the PSTN provider and let investigate into this case.

3. Check the VoIP Switch's emergency routing:

◊ Emergency numbers
◊ Emergency number rewriter
◊ Routing Tables toward the PSTN
◊ RuleSet that may tag outgoing calls toward emergency numbers

4. Check if any IP network devices may interfere with the SIP signaling:

◊ If there are external Session Board Controller SBC or SIP-SS7 Gateway involved check their behavior concerning the emergency calls
◊ If a firewall FW is involved check that no SIP ALG or "SIP Helpers" are active

5. Treat the problem:

a) Adjust the emergency routing of the VoIP Switch if needed

b) Fix the IP network devices if needed

6. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to ServiceCenter Message "TopStop"

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> msgs ServiceRatingControl (01) <CALLING_NUMBER> "max available charges reached for account:"`
`<HOST_NAME> msgs AlarmLogger "[TOPSTOP][ALARM] tenant" <TENANT> "topstop limit nearly reached for account"`

Description:
A user's TopStop limit was reached!

Note	A TopStop alarm early in the month or for a lot of users indicates a possible fraud case!

Consequences:

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ A TopStop alarm early in the month indicates a possible fraud case

For the user:

◊ No outgoing calls except emergency call will work when the TopStop limit is reached

Solution:
If it is a regular TopStop then contact the user and enhance the monthly TopStop limit.

If it is a fraud situation handle according "Best Practice: Fraud"

Action:
1. Check if it is a regular TopStop situation.

2. Check if it is a possible fraud case:

◊ Reached TopStop limit early in the month?
◊ Concurrently a lot of TopStop limits reached?
◊ High call peak during the night or weekend?
Check at Xymon Column " calls_sys " .

3. Treat according " Best Practice for "Fraud Situation"

4. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Nimbus Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> msgs "NimbusLink (ue) Cannot subscribe"`

Description:
The Nimbus component is a VoIP Switch internal bus that connects the various VoIP Switch components on the servers. If a Nimbus endpoint on one server is missing the other Nimbus endpoints start to complain.

If a Nimbus endpoint is missing then the component may be stopped, the server not on line or an IP network problem.

This error is often displayed during VoIP Switch software upgrades of the servers. In this situation just wait until the upgrade is finished.

Consequences:

Warning	This erroneous condition must be checked and treated within reasonable time!

For the VoIP Switch telephony service:

◊ Usually none

For the operations:

◊ None

For the user:

◊ None

Solution:
Solve the IP network problems or server problems if needed.

Action:
1. Check if the IP network is OK

2. Check the status of the VoIP Switch components located on the server where the Nimbus is missing:

Is only Nimbus missing or other components to on this server?

3. Treat the problem:

a) If there are IP network problems

Actions see: "Maintenance Due to IP Network Alarm"

b) If there is not a planned outage then try to solve the server problem

c) If there is not a planned outage then try to restart the Nimbus on this server:

```
root# nimbus restart
```

4. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Messages from CallAgent Server

Maintenance Due to CallAgent Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> msgs

Description:
The CallAgent treats the message exchange with the MGCP MTA. The CallAgent has an all active redundancy scheme. If one CallAgent fails the remaining CallAgent take over the work load.

Consequences:

Warning	This erroneous condition must be checked within short time!

For the VoIP Switch telephony service:

◊ As long one CallAgent remains the VOIP Switch works

For the operations:

◊ None

For the user:

◊ Single MGCP MTA at the user's premises is not working correctly. The telephone service may not always work for this users.

Solution:
Depends on the analyzed problem.

Action:
1. Check if the IP network is OK

2. Check the status of the CallAgent components

Confirm that the reported CallAgent server is affected

3. Check the reported CallAgent server with the "Server Administrator (OMSA)"

4. Treat the problem:

a) If there are IP network problems

Actions see: "Maintenance Due to IP Network Alarm"

b) If there is a CallAgent problem try to restart the component:

```
root# callagent restart
```

5. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Messages from CPECenter Server

Maintenance Due to CpeCenterMessage

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> msgs
<HOST_NAME> msgs "DevAdmProvider (-1) duplicated devicetype:" <DEVICE_TYPE>

Description:
During the preparation of a device configuration file two device configuration templates were found. If a CPE loads a device configuration file which was produced under these conditions it may not work correctly.

Consequences:

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ None

For the user:

◊ The CPE may not work with the produced configuration file

Solution:
One device configuration template has to be deleted.

Action:
1. Contact the "VoIP Switch Supplier Support"!

Maintenance Due to IP Network Alarms

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> conn "Host does not respond to ping"
<IP_ADDRESS>
```

Description:
This test performs a "ping" toward the IP address of the host. If the "ping" is not answered then there is a problem with the IP network, e.g.:

◊ Pinged host defect or off line
◊ Layer2 IP Switch defect or off line
◊ Brocken IP backbone network

Consequences:

Warning **MOST SEVERE** condition if several VoIP Switch server are affected for a longer duration (ca 15min)!

For the VoIP Switch telephony service:

◊ The telephone service may be interrupted

For the operations:

◊ The MySQL databases may loose their replication

For the user:

◊ The telephone service may be interrupted for the users!

Solution:
Solve the IP network problems!

Check the IP network devices:

◊ Pinged host
◊ Layer 2 IP switches
◊ IP Routes
◊ Firewalls

Check the VoIP Switch server IP connectivity.

Action:
1. Evaluate the severity of the IP network outage:

a) Check if it is a occasional ping failure:

- Only one host doesn't respond
- Only 1 or 2 poll cycle fail
  Type "Occasional Failure":
- In this situation the erroneous situation may be neglected.

b) Check if it is only a single host:

- One host doesn't respond anymore
  Type "Host Failure":
  - ◊ Check the hardware condition and IP connectivity of this device
  - ◊ Check with the VoIP Switch Administrator in the ConfigCenter Menu "Components" how the VoIP Switch is affected

c) Check if more than one VoIP Switch server is affected:

- More than one VoIP Switch server don't respond anymore
  Type "VoIP Switch Failure":

1. Check with the VoIP Switch Administrator how the VoIP Switch is affected:

   a) Connect to both (*-ms-01, *-ms-02) ConfigCenter Menu "Components" and check the component status

   b) Check the questions:

   ◊ Which VoIP Switch servers are not visible?
   ◊ Are they the same on both ConfigCenter?

   ◊ Does one ConfigCenter see only the servers on its side? E.g.:
   Side A components complain that they doesn't see their peers on Side B?
   Side B components complain that they doesn't see their peers on Side A?
   If yes => There is a heavy IP backbone problem

   c) Check in the ConfigCenter Menu **Channles** if new connections were established since the IP outage

   If yes => Some users still can make phone calls

2. Check with the VoIP Switch Administrator how the users are affected:

   a) Connect to both (*-ms-01, *-ms-02) Xymon Column "regs" and check the CPE and MTA registrations status.

   b) Check the questions:

   ◊ Check: Do drop the user's CPE registration?
   If yes => There is a heavy IP backbone problem some users cannot use the telephony service anymore!

3. Treat the Type "VoIP Switch Failure":

   a) VoIP Switch Administrator:

   In this situation the erroneous situation may be neglected. Observe if the situation remains.

2. Treat the Type " Occasional Failure ":

   a) VoIP Switch Administrator:

   If possible pre-bar the VoIP Switch component on this server

   b) Solve the IP or hardware issue with the failed host

3. Treat the Type "VoIP Switch Failure":

   a) VoIP Switch Administrator:

   Contact the "VoIP Switch Supplier Support"

4. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Operating System Alarms

The VoIP Switch Administrator and/or server service personnel find here instructions for managing problems indicated by the operating system supervision.

Maintenance Due to Supervised Processes Missing

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> procs "Processes not OK" <MISSING_PROCESS>
```

Description:
One or more supervised process of a Linux service or VoIP Switch component is missing.

Consequences:

Warning **SEVERE erroneous condition** that must be handled!

For the VoIP Switch telephony service:

- ◊ Depends If a VoIP Switch component is missing then the VoIP Switch looses redundancy capability
- ◊ If a Linux service is missing the VoIP Switch may be hampered or the server is not working correctly

For the operations:

- ◊ Depends on the VoIP Switch components or Linux service

For the user:

- ◊ Depends on the VoIP Switch components or Linux service

Solution:
Restart the VoIP Switch component or Linux service.

Action:
1. Check with the VoIP Switch Administrator if it is possible to restart the component or service without endangering the VoIP Switch telephony service.

If possible pre-bar the VoIP Switch component via the ConfigCenter!

2. Restart the VoIP Switch component or Linux service:

a) Restart the VoIP Switch component

```
root# <COMPONENT> restart
```

◊ Example:

```
root# servicecenter restart
```

b) Restart the service:

```
root# /etc/init.d/<SERVICE> restart
```

◊ Example:

```
root# monit restart
```

3. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Supervised IP Ports

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> ports "Ports not OK" <MISSING_PROCESS_PORTS>
```

Description:
One or more supervised IP port of a Linux service or VoIP Switch component is missing.

Consequences:

Warning **SEVERE erroneous condition** that must be handled!

For the VoIP Switch telephony service:

◊ Depends If a VoIP Switch component is missing then the VoIP Switch looses redundancy capability
◊ If a Linux service is missing the VoIP Switch may be hampered or the server is not working correctly

For the operations:

◊ Depends on the VoIP Switch components or Linux service

For the user:

◊ Depends on the VoIP Switch components or Linux service

Solution:
Restart the VoIP Switch component or Linux service.

Action:
1. Check with the VoIP Switch Administrator if it is possible to restart the component or service without endangering the VoIP Switch telephony service.

If possible pre-bar the VoIP Switch component via the ConfigCenter!

2. Restart the VoIP Switch component or Linux service:

a) Restart the VoIP Switch component

```
root# <COMPONENT> restart
```

◊ Example:

```
root# servicecenter restart
```

b) Restart the service:

```
root# /etc/init.d/<SERVICE> restart
```

◊ Example:

```
root# monit restart
```

3. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Supervised Hard-Disk Usage

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> disk "File systems not OK"
```

Description:
A hard-disk or hard-disk partition is full. If a hard-disk is full then the Linux operating system behaves unpredictable and the server will most probably crash.

Consequences:

Warning **SEVERE erroneous condition** that must be handled!

For the VoIP Switch telephony service:

◊ Depends on the VoIP Switch components running on the server

For the operations:

◊ Depends on the VoIP Switch components running on the server

For the user:

◊ Depends on the VoIP Switch components running on the server

Solution:
Identify big files or directories. Delete or archive files externally.

Action:
1. Check hard-disk usage:

```
root# df -h
```

2. Find fat files:

```
root# ls -lahS $(find / -type f -size +100k)
```

◊ Example find file sizes >60MByte:

```
root# ls -lahS $(find /opt/backup/ -type f -size +60000k)
```

◊ Check for fat files in the following suspicious directories:
/opt/backup/

◊ Do not touch big files in:
/var/lib/mysql/

3. Find big directories:

```
root# du -hs
```

Example of a more specific search find directory sizes >1GByte:

```
root# du -hs /home/ratingcenter/* | grep G
root# du -hs /home/*/* | grep G
```

◊ Check the following suspicious directories:
/opt/backup/
/home/mediacenter/messages
//home/ratingcenter/cdrs

4. Prior of deleting files or directories check with the VoIP Switch Administrator if they are not needed anymore!

If you are suspicious but not sure if it is wise to delete a certain file or directory then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Supervised Memory Usage

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> memory "Memory low"

Description:
One or more processes consume a lot of memory space. If the memory becomes low the operating system Linux start to swap memory to and from hard-disk. This reduces the performance of the server.

Consequences:

Warning	This erroneous condition must be handled within reasonable time!

For the VoIP Switch telephony service:

◊ Depends on the VoIP Switch components running on the server

For the operations:

◊ None

For the user:

◊ None

Solution:
Identify which process or consumes the memory. Restart the process in order to free memory. Stop and restart the swapping on the server.

Action:

1. If a LoadBalancer *-lb-* or ServiceCenter *-sc-* server is affected:

   Contact the "VoIP Switch Supplier Support"!

2. Find which processes use the memory:

   ◊ This is a difficult task!

```
root# top
```

3. Stop and restart the swapping:

   Preconditions:

   ◊ Choose a day time where the server is not in high load.
   ◊ If possible pre-bar the VoIP Switch components on this server via the ConfigCenter
   ◊ Make sure that the redundant VoIP Switch component is running

   a) Restart the responsible process:

```
root# /etc/init.d/<PROCESS_NAME> restart
```

   b) Stop the swapping:

   ◊ Don't do this during high load!
   ◊ It will take some time until accomplished!

```
root# swapoff -a
```

   c) Restart the swapping:

```
root# swapon -a
```

   d) Check if the swap is working regularly:

```
root# swapon -s
```

Maintenance Due to Supervised CPU Load

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> cpu "Load is High"

Description:
One or more processes consume extensively CPU power. This may reduce the performance of the server.

Consequences:

Warning	This erroneous condition must be handled within reasonable time!

For the VoIP Switch telephony service:

◊ Reduced performance on the affected server and VoIP Switch component

For the operations:

◊ None

For the user:

◊ None

Solution:
The CPU consuming process has to be identified. If a process is identified it has to be checked if it is a regular or erroneous situation.

If it is a regular situation then it has to be investigated if the servers computing power is still sufficient for this VoIP Switch. If the server hosts a VoIP Switch component which offers an configurable load acceptance via the ConfigCenter then it is worth a try to reduce the components workload.

An erroneous situation can mostly be solved by restarting the process.

Action:
1. Identify the responsible process:

a) Check the process situation with:

```
root# top
root# ps aux
```

b) If a process is suspicious check for multiple processes of the same name:

```
root# ps -aef
```

c) If a process is suspicious check for zombie processes (lists the zombie process id):

```
root# ps aux
```

d) Evaluate with the VoIP Switch Administrator if the suspicious process is in a regular or erroneous state.

2. Handle an erroneous Linux process state.

a)* Restart a Linux process:

```
root# /etc/init.d/<PROCESS_NAME> restart
```

b) Kill a process, e.g. double started process, zombie:

```
root# kill -9 <PROCESS_ID>
```

3. Handle a VoIP Switch component :

a) Restart an erroneous VoIP Switch component:

```
root# <COMPONENT_NAME> restart
```

b) If the VoIP components ServiceCenter or MediaServer produces high load then the VoIP Switch Administrator may reduce their accepted work load via the ConfigCenter.

4. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Supervised Files Missing or to Big

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> ????

Description:

Consequences:

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ None

For the user:

◊ None

Solution:

Action:
1. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

VoIP System Maintenance

Best Practice for Handling a "Fraud" Situation

The Aarenet VoIP Switch Administrator finds here instructions for managing fraud problems.

1. Immediate action:

◊ Block call routing to the destination (usually somewhere in the Caribbean, west or central Africa)
◊ If only from one source IP address then block this IP address on the FW

2. Investigate if the fraud is due to "Direct Registrations" with correct SIP credentials on the VoIP Switch:

◊ Check if the calling number has multiple SIP registrations of a suspicious source IP range or user agent!
If YES then:
The SIP credentials were not kept secret or hacked from the users CPE

Action:

- Block this user account for outgoing calls (blocking international calls is usually sufficient)
- Change the SIP credential in the user account and the user's CPE.
- Change the CPE administration login credentials

3. Investigate if the fraud is due to "Hacked Users CPE":

a) Analyze the traces of some fraud connections.

Check if the source IP remain the one of a registered user CPE!
If YES then:
If yes block this user account for outgoing calls

Action:

- Block this user account for outgoing calls (blocking international calls is usually sufficient)
- Inform the user about the fraud and its reason
- Change the SIP credential in the user account and the user's CPE.
- Change the CPE administration login credentials

4. Post Work:

◊ Undo the "immediate action"
◊ Enable the customer account when the SIP credentials and CPE administration login credentials are changed

```
ar:both" />
```

Put Back to Service a VoIP Switch Component

There are two ways to put back to service a VoIP Switch Component:

Variant 1: "Start it"

Action:

A) Start and check the component via the shell:

```
root# <AS_COMPONENT> start
root# <AS_COMPONENT> status
```

The consequence is that the component starts immediately its operative work.

Variant 2: "Start it gracefully"

This variant may make sense when the following VoIP Switch components shall become active but not operative immediately:

◊ ServiceCenter
◊ MediaServer

◊ FaxServer
◊ CallAgent

Action:

A) Start "passive" the component via the ConfigCenter.

```
root# <AS_COMPONENT> startpassive
root# <AS_COMPONENT> status
```

B) Make the component operative at the appropriate time:

```
ConfigCenter GUI   Menu "System"    Menu "Components"

         Click the desired VoIP Switch component

           Change the parameter "Acceptance" to 100
```

The "Acceptance" may by any value >0 according. Choose according the load balancing scheme of the component.

C) Check if the component displays activity:

```
ConfigCenter GUI   Menu "System"    Menu "Components"
```

Work Flow for Analyzing VoIP Switch Problems

Note	Not every red alarm jeopardizes the telephony service as a whole but a bulk of yellow warnings may endanger it!

The VoIP Switch Administrator and other service personnel find here a work flow for analyzing VoIP Switch problem indications and find out the appropriate action.

The main task is to find out if:

1. The situation jeopardizes the telephony service as a whole, e.g.:
   · IP network issues
   · Several VoIP Switch servers failed or off line
2. The database replication is broken
   · IP network issues
   · Server with running database failed
   · Linux service MySQL failed
3. The situation hampers the operation of configuration of customer accounts, addresses etc.
   · Management server failed or off line
   · VoIP Switch component ConfigCenter, AdminCenter DataAccessCenter, RatingCenter stopped working correctly

The VoIP Switch Administrator finds here the work flow for analyzing VoIP Switch problems:

Work Flow for Analyzing VoIP Switch Problems

Analysis:

1. Check if it is a **single alarm** or a **bulk alarm** situation.

   a) Connect to the VoIP Switch monitor Xymon "Main View"

      As a rule of thumb: It is a single error if only one issue is displayed.

2. Analyze and treat a **single alarm** situation:

a) Check the contents of the error message.

b) Compare the error description against the **Indication "Xymon Event"** ones in chapter "VoIP Switch Maintenance"

c) Check if the actual situation is equal or similar as described and the recommended actions suitable.

d) Execute the suitable actions.

If you are not sure contact the "VoIP Switch Supplier Support"

3. Analyze the **bulk alarm** situation:

a) Get a first overview of the situation by analyzing the Xymon Monitor :

Check in the MS-01 Xymon monitor the server, component and IP status:

```
Xymon GUI    Xymon "Main View"
```

1. Which type of server are affected?
   - At least one LoadBalancer LB server must be active that the telephony service can work!
   - At least one ServiceCenter SC server must be active that the telephony service can work!
   - At least one server with the operative database must be active that the telephony service can work!
2. Check the CPE registration statistic :
   - Do drop the CPE registrations?
3. Check the call statistic:
   - Do drop the VoIP Switch number of calls?

     ```
     Xymon GUI   Management Server   Column "calls_sys"
     ```
   - Do drop the calls on one or more ServiceCenter?

     ```
     Xymon GUI   ServiceCenter Server   Column "calls_sc"
     ```
   - Do drop the calls on one or more gateways?

     ```
     Xymon GUI   Gateway   Column "calls_gw"
     ```
4. Do the same check as above on MS-02 Xymon Monitor
5. Does the comparison of the two Xymon Monitor point out that:
   - The same single component on the same server failed?
   - All components of one side failed?
   - The Xymon Monitor sees only the components on its side?
   - The telephony service is running at least on one side

b) Extend the overview by analyzing the ConfigCenter "System Component" Overview :

Check in the MS-01 ConfigCenter the status of the VoIP Switch components:

```
ConfigCenter GUI   Menu "System"    Menu "Components"
```

1. Are actually calls running and new calls can be established?
2. Make test calls:
   - To and from a telephone number in the PSTN
   - On-net test calls
   - Call a well known VoiceMail Box from on-net and from PSTN
3. Is the number of running calls fast dropping and no new calls are established?
4. Which type of VoIP Switch components are affected?
   - At least one LoadBalancer component must be active that the telephony service can work!
   - At least one ServiceCenter component must be active that the telephony service can work!
   - At least one operative database must be active that the telephony service can work!
   - Does this picture correspond to the results of the first overview in the Xymon Monitor ?
5. Do the same check as above on MS-02 ConfigCenter

6. Does the comparison of the two ConfigCenter point out that:
   - ♦ The same single component on the same server failed?
   - ♦ All components of one side failed?
   - ♦ The ConfigCenter sees only the components on its side?
   - ♦ The telephony service is running at least on one side

4) Treat **bulk alarm** situations:

a) Is there a VoIP Switch server hardware, RAID or hard-disk problem?

Indications:

Indication:
<HOST_NAME> "snmptrapd" "failure"
<HOST_NAME> "snmptrapd" "degraded"

Actions:

For DELL server see: "Treating Problems of Servers from DELL Inc ®"

b) Is the IP connectivity affected to or between VoIP Switch servers?

Note	If VoIP Switch servers are affected then a lot of additional alarming messages of missing VoIP Switch components will pop up!! **This can be one of the most annoying erroneous situations!**

Indications:

Indication:
<HOST_NAME> conn "Host does not respond to ping" <IP_ADDRESS>
* Dropping CPE registrations ! • Dropping calls ! • No new calls!

Actions:

See: "Maintenance Due to IP Network Alarm"

c) If you are not sure what to do then contact the "VoIP Switch Supplier Support"

VoIP Switch Server Maintenance

Maintenance Due to VoIP Switch Components General Alarms

Maintenance Due to Messages from Java Framework

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> msgs "Jdbc"`

Description:
Java internal exceptions. Mostly due to database accesses which are hopefully handled by the application.

Consequences:
For the VoIP Switch telephony service:

◊ Mostly none

For the operations:

◊ Mostly none

For the user:

◊ Mostly none

Solution:
Observe the frequency of this event

Action:
1. Observe the frequency of this event

2. If the erroneous condition is to frequent then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Messages from VoIP Switch Components Internals

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> msgs "EventQueue"`
`<HOST_NAME> msgs "SysCompDatabase - Cannot evalute status"`

Description:
These events may happen on all VoIP Switch servers and are VoIP Switch component internal notes.

Consequences:
For the VoIP Switch telephony service:

◊ Mostly none

For the operations:

◊ Mostly none

For the user:

◊ Mostly none

Solution:
Observe the frequency of this event

Action:
1. Observe the frequency of this event

2. If the erroneous condition is to frequent then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Messages from LoadBalancer Server

Maintenance Due to HealthCheck Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> msgs "HealthCheck"`

Description:
The HealthCheck supervises the status of virtual IP addresses and their associated physical IP addresses. If the HealthCheck on one server doesn't see the peer physical IP address it takes over the virtual IP address. It most probably points out an IP network problem in the "Public Voice Segment"

Consequences:

Warning	This erroneous condition must be checked within reasonable time!

For the VoIP Switch telephony service:

◊ None if concurrently no other IP network problems arise

For the operations:

◊ None

For the user:

◊ None

Solution:
Solve the IP network if needed.

Check status the VoIP Switch component with an active-passive scheme:

◊ LoadBalancer
◊ CallBalancer
◊ RatingCenter

Action:
1. Check if the IP network is OK

2. Check the status of the LoadBalancer components

Confirm if the active LoadBalancer swapped, e.g. from *-lb-01 to *-lb-02

3. Check the status of the CallBalancer components

Confirm if the active CallBalancer swapped, e.g. from *-lb-01 to *-lb-02

4. Check the status of the RatingCenter components

Confirm if the active CallBalancer swapped, e.g. from *-ms-01 to *-ms-02
Confirm if the active RatingCenter is processing the CDR's

5. Treat the problem:

a) If there are IP network problems

Actions see: "Maintenance Due to IP Network Alarm"

b) If there is a LoadBalancer problem try to restart the component:

```
root# loadbalancer restart
```

c) If there is a CallBalancer problem try to restart the component:

```
root# callbalancer restart
```

d) If there is a RatingCenter problem try to restart the component:

```
root# ratingcenter restart
```

e) If the RatingCenter swapped make sure that the CDR are processed:

1. ConfigCenter GUI Menu "System" Menu "Components"
Click line at "active" RatingCenter -> In dialog select "Process CDRs"
Click button [Close]
2. The CDR CSV-Files are processed:

```
root# cd /home/servicecenter/cdrs
```

Check if the CSV files have an actual time stamp which indicates that new CDRs where written:

```
root# ls -ltra
```

Open a CSV file and check for new entries, e.g.:

```
root# less monthly.csv
```

6. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

If those events are logged subsequently then rapport it to the "VoIP Switch Supplier Support"!

Maintenance Due to LoadBalancer Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> msgs "Balancer"
```

Description:
LoadBalancer internal problem that is treated internally by the component. The LoadBalancer has an "active-passive" redundancy scheme.

Consequences:

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ None

For the user:

◊ None

Solution:
Not defined yet

Action:
1. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

If those events are logged subsequently then rapport it to the "VoIP Switch Supplier Support"!

Maintenance Due to LoadBalancer Message "Missing ServiceCenter"

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> msgs "BalancerSwitch" <SERVICECENTER> "not
available anymore"
```

Description:
The LoadBalancer indicates that it doesn't see a certain ServiceCenter.

This happens when:

◊ the ServiceCenter has restarted
the event will be transient
◊ the ServiceCenter is stopped
the event will remain until the ServiceCenter is started again
◊ no IP connectivity
the event will remain until the IP connectivity is reestablished

Consequences:

Warning	This erroneous condition must be handled within reasonable time!

For the VoIP Switch telephony service:

◊ None, the other ServiceCenter take over the work load
◊ If a ServiceCenter is missing then the VoIP Switch looses redundancy capability

For the operations:

◊ None

For the user:

◊ None

Solution:
Solve the IP network problems if needed:

Actions see: "Maintenance Due to IP Network Alarm"

Solve the server problem if needed

Actions see: "Treating Server Hardware Problems"

Action:
1. Check if the IP network is OK

2. Check the status of the ServiceCenter components

Confirm that the reported ServiceCenter server is affected

3. Check the reported ServiceCenter server with the "Server Administrator (OMSA)"

4. Treat the problem:

a) If there are IP network problems

Actions see: "Maintenance Due to IP Network Alarm"

b) If there is a ServiceCenter problem try to restart the component:

```
root# servicecenter restart
```

5. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to CallBalancer Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> msgs

Description:
The CallAgent dispatches MGCP messages to the CallAgent components.

The CallAgent has an "active-passive" redundancy scheme.

Consequences:

Warning	This erroneous condition must be checked within short time!

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ None

For the user:

◊ Users with MGCP MTA as telephone adapter may not be able to telephone

Solution:
Check status the CallBalancer active-passive scheme and if the MGCP messages are processed.

Action:
1. Check if the IP network is OK

2. Check the status of the CallBalancer components:

a) Confirm if the active CallBalancer swapped , e.g. from *-ms-01 to *-ms-02

b) Confirm if the active CallBalancer is processing the MGCP messages

Check if the CallAgent treat MGCP connections and that the total number of MGCP connections is not dropping.

3. Check if the MGCP audits are not dropping:

a) Connect to a Xymon monitor and check in Xymon Column "regs" the numbers of MGCP-Active and MGCP-Brocken

b) Check the questions:

◊ Do drop the number of MGCP-Active?
If yes => There may be a IP backbone problem or CallBalancer, CallAgent outage!

4. Treat the problem:

a) If there are IP network problems

Actions see: "Maintenance Due to IP Network Alarm"

b) If there is a CallBalancer problem try to restart the component:

```
root# callbalancer restart
```

5. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to MediaServer Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> msgs "MediaConnection (06) Cannot handle outgoing message"`
`<HOST_NAME> msgs "MediaServerProvider (MS) refreshing mediaserver mc1ms2 failed"`

Description:
The MediaServer records or plays back announcements and VoiceMail messages. Occasionally it may not correctly record a message and transfer it to the MediaCenter or play back an announcement or message.

The MediaServer can act as media proxy for active connections and transcode media streams.

Consequences:

Warning	If in this VoIP Switch the MediaServer acts as media proxy then the erroneous situation must be checked soon!

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ None

For the user:

◊ A VoiceMail Box message or announcement couldn't correctly record or played back.
◊ User may not hear the other side or vica versa.

Solution:
Depends on the situation.

Action:
1. If the erroneous condition remains or happens to often then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Messages from Management Server

Maintenance Due to AdminCenter Message "Missing FMC Application Server"

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> msgs "FmcRequest - Cannot post request"`
`<HOST_NAME> msgs "FmcProvider - could not provision pbx"`

Description:
The AdminCenter tried to configure the FMC application.

Consequences:

Warning	This erroneous condition is sporadic or must be handled within reasonable time!

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ A configuration on a FMC server failed

For the user:

◊ A user "an MC-Phone" is not working

Solution:
Check the state of the FMC servers and their IP connectivity toward the VoIP Switch servers.

Action:
1. Check if the IP network is OK

2. Check the status of the FMC server

3. Treat the problem:

a) If there are IP network problems

Actions see: "Maintenance Due to IP Network Alarm"

b) If there is a FMC server problem

Contact the "VoIP Switch Supplier Support"!

4. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to AdminCenter Message "Missing Redirection Server"

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> msgs "FmcProvider - could not provision user"
<USER_TELEPHONE_NUMBER>
```

Description:
The mobile app "an MC-Phone" couldn't get the information from the associated redirection server (by default a Comdasys server located in Europe) where its responsible configuration server is located. Therefore the users "an MC-Phone" couldn't obtain its configuration and will not work.

Consequences:

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ None

For the user:

◊ The mobile app "an MC-Phone" will not work

Solution:
Make sure to have good IP connectivity to the Internet

Action:
1. The user must find a reliable Internet connection and restart the app "an MC-Phone" until it gets its configuration

Maintenance Due to ConfigCenter Message "Wrong User Login"

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> msgs "msgsAccessLogger - ADMIN:login; user"
<USERNAME> "-> User Blocked"
```

Description:
A VoIP Switch Administrator, Operator, Supporter tried to login to the ConfigCenter with wrong credentials. The user will be blocked for several minutes.

Consequences:

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ The user will be blocked from the ConfigCenter for several minutes.

For the user:

◊ None

Solution:
Wait

Action:
1. Retry after a few minutes with the correct login credentials.

2. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to ConfigCenter Message "DB Replication Check"

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> msgs JdbcReplicationMonitor "Replication"
'<BROKEN_REPLICATION_DIRECTION>' "is broken!"
```

Description:
The database replication check was not successful. This can happen from time to time when the database has to process heavy load.

In most cases the database replication recovers automatically even after several hours of failed replication. If it is not recovering then this is a severe problem and must be treated.

Consequences:

Warning	If this erroneous condition remains then this is a SEVERE erroneous condition and must be treated within short time!

For the VoIP Switch telephony service:

◊ The database redundancy is endangered

For the operations:

◊ None

For the user:

◊ None

Solution:

Restore the MySQl DB replication if the erroneous condition remains.

Action:
1. Check periodically (ca. every half hour) the Xymon monitor for this error condition.

2. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to DataAccessCenter Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> msgs "Jdbc" "SQL-Exception during statement"`

Description:
A configuration via the DataAccessCenter may have failed.

This may happen if the database is under heavy load.

Consequences:

Warning	This erroneous condition must be checked within reasonable time!

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ A customer configuration may have failed (which is hopefully covered by the CRM application).

For the user:

◊ None

Solution:
Inter-working between the DataAccessCenter and database must be optimized.

Action:
1. If this Java event is logged subsequently then rapport it to the "VoIP Switch Supplier Support"!

Maintenance Due to RatingCenter Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> msgs`

Description:
The RatingCenter has an "active-passive" scheme. Every RatingCenter event has to be checked if the active RatingCenter is working correctly and is processing the CDRs.

Consequences:

Warning	This erroneous condition must be checked within short time!

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ A CDR may be not written correctly into the CDR database and/or CSV files.
◊ The customer billing contains not all CDR

For the user:

◊ None

Solution:
Check status the RatingCenter active-passive scheme and if the CDR are processed.

Action:
1. Check the status of the RatingCenter component

Confirm if the active RatingCenter is processing the CDR's

2. Treat the problem:

a) If the RatingCenter swapped make sure that the CDR are processed:

Open the ConfigCenter Menu "Components"

Click line at "active" RatingCenter -> In dialog select "Process CDRs"

Click button [Close]

b) Check if the CDR CSV-Files are processed:

Open the CDR directory:

```
root# cd /home/ratingcenter/cdrs
```

Check if the CSV files have an actual time stamp which indicates that new CDRs where written:

```
root# ls -ltra
```

Open a CSV file and check for new entries, e.g.:

```
root# less monthly.csv
```

3. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Messages from ServiceCenter Server

Maintenance Due to FaxServer Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> msgs

Description:
Fax may not received correctly. The mailing of the PDF file may fail.

Consequences:

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ None

For the user:

◊ A received Fax may not be correctly received and transferred to the user. This situation is usually handled by the Fax device either automatically or manually.

Solution:
Restart the FaxServer component.

Action:
1. Check if no Fax at all are received.

Send test fax.

2. Restart the FaxServer:

```
root# faxserver restart
```

3. If the FaxServer logs subsequently then rapport it to the "VoIP Switch Supplier Support"!

Maintenance Due to MediaCenter Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> msgs MediaCenterCall
<HOST_NAME> msgs MediaServer
<HOST_NAME> msgs "file not found"

Description:
The MediaCenter handles the WAV files from announcements and VoiceMail messages. Occasionally it may not correctly record a message, loose a message file. Also an order to the MediaServer may fail to replay a message or announcement.

Consequences:

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ None

For the user:

◊ A VoiceMail Box message or announcement couldn't correctly recorded or played back

Solution:
Clean up the VioceMail message date base.

Optimize the inter-working of MediaCenter and MediaServer

Action:
1. If those events are logged subsequently then rapport it to the "VoIP Switch Supplier Support"!

Maintenance Due to ServiceCenter Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> msgs`

Description:
The ServiceCenter is the main component of the VoIP Switch. It computes the connections signaling and telephony features.

The ServiceCenter has an all active redundancy scheme. If one ServiceCenter fails the remaining ServiceCenter take over the work load.

Consequences:

Warning	This erroneous condition must be checked and treated within short time!

For the VoIP Switch telephony service:

◊ As long one ServiceCenter remains the VoIP Switch works!

For the operations:

◊ None

For the user:

◊ None

Solution:
Depends on the analyzed problem.

Action:
1. Check how acute the problem is:

a) Check if the IP network is OK

b) Check the status of the ServiceCenter component

◊ Are enough ServiceCenter active that the work load can be treated?
If NO then there is a most SEVERE erroneous situation

c) Check in the ConfigCenter Menu "Components" if the active ServiceCenter is processing the connections:

◊ Do drop the total number of connections?
If YES then there is a most SEVERE erroneous situation:
There may be a IP backbone problem!

d) Check in the Xymon Column "regs" the number of registered SIP-Devices:

◊ Do drop the number of SIP-Devices?
If YES then there is a most SEVERE erroneous situation:
There may be a IP backbone problem!

e) Check the reported ServiceCenter server with the "Server Administrator (OMSA)"

◊ Are problems signaled?

2. Treat the problem:

a) If there are IP network problems

Actions see: "Maintenance Due to IP Network Alarm"

b) If there is a ServiceCenter problem try to restart the component:

```
root# servicecenter restart
```

c) If there is a hardware problem:
Actions see: "Treating Server Hardware Problems"

3. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to ServiceCenter Message "License Violation"

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> msgs License "License Violation"
<HOST_NAME> msgs License "grace-period remaining:"

Description:
This ServiceCenter has a license problem and will work only for the remaining grace period.

Consequences:

Warning	This erroneous condition must be checked and treated within the remaining grace period!

For the VoIP Switch telephony service:

◊ As long one ServiceCenter remains the VOIP Switch works
◊ The telephony service will be stopped on this ServiceCenter after passing of the grace period

For the operations:

◊ None

For the user:

◊ None

Solution:
Get actual licenses from the VoIP Switch Supplier.

Action:
1. Check in the ConfigCenter Menu "Components" which ServiceCenter component has a license problem and how long the grace period is.

2. Contact the "VoIP Switch Supplier Support"!

Maintenance Due to ServiceCenter Message "Failed Emergency Call"

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> msgs ServicePrioCallControl "Could not establish priority-call". Call from Connection/<SIP_CALL_ID>/<CALLING_NUMBER> to <CALLED_EMERGENCY_NUMBER>`

Description:
A user's emergency call failed!

Consequences:

Warning	**Severe** legal condition that must be handled! This case can have legal consequences for the provider!

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ None

For the user:

◊ The emergency call did not work

Solution:
Check if the call routing failed due to a VoIP Switch emergency call treating or routing. If yes fix them.

Check if the PSTN provider did reject the emergency call. If yes contact the PSTN provider.

Action:
1. Archive traces for legal responsibilities:

◊ Save the trace of this emergency call and all subsequent calls from this user toward emergency numbers

2. Check where the call was rejected.

◊ If the call was rejected at the PSTN provider side contact the PSTN provider and let investigate into this case.

3. Check the VoIP Switch's emergency routing:

◊ Emergency numbers
◊ Emergency number rewriter
◊ Routing Tables toward the PSTN
◊ RuleSet that may tag outgoing calls toward emergency numbers

4. Check if any IP network devices may interfere with the SIP signaling:

◊ If there are external Session Board Controller SBC or SIP-SS7 Gateway involved check their behavior concerning the emergency calls
◊ If a firewall FW is involved check that no SIP ALG or "SIP Helpers" are active

5. Treat the problem:

a) Adjust the emergency routing of the VoIP Switch if needed

b) Fix the IP network devices if needed

6. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to ServiceCenter Message "TopStop"

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> msgs ServiceRatingControl (01) <CALLING_NUMBER> "max available charges reached for account:"`
`<HOST_NAME> msgs AlarmLogger "[TOPSTOP][ALARM] tenant" <TENANT> "topstop limit nearly reached for account"`

Description:
A user's TopStop limit was reached!

Note	A TopStop alarm early in the month or for a lot of users indicates a possible fraud case!

Consequences:

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ A TopStop alarm early in the month indicates a possible fraud case

For the user:

◊ No outgoing calls except emergency call will work when the TopStop limit is reached

Solution:
If it is a regular TopStop then contact the user and enhance the monthly TopStop limit.

If it is a fraud situation handle according "Best Practice: Fraud"

Action:
1. Check if it is a regular TopStop situation.

2. Check if it is a possible fraud case:

◊ Reached TopStop limit early in the month?
◊ Concurrently a lot of TopStop limits reached?
◊ High call peak during the night or weekend?
Check at Xymon Column " calls_sys " .

3. Treat according " Best Practice for "Fraud Situation"

4. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Nimbus Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> msgs "NimbusLink (ue) Cannot subscribe"
```

Description:
The Nimbus component is a VoIP Switch internal bus that connects the various VoIP Switch components on the servers. If a Nimbus endpoint on one server is missing the other Nimbus endpoints start to complain.

If a Nimbus endpoint is missing then the component may be stopped, the server not on line or an IP network problem.

This error is often displayed during VoIP Switch software upgrades of the servers. In this situation just wait until the upgrade is finished.

Consequences:

Warning	This erroneous condition must be checked and treated within reasonable time!

For the VoIP Switch telephony service:

◊ Usually none

For the operations:

◊ None

For the user:

◊ None

Solution:
Solve the IP network problems or server problems if needed.

Action:
1. Check if the IP network is OK

2. Check the status of the VoIP Switch components located on the server where the Nimbus is missing:

Is only Nimbus missing or other components to on this server?

3. Treat the problem:

a) If there are IP network problems

Actions see: "Maintenance Due to IP Network Alarm"

b) If there is not a planned outage then try to solve the server problem

c) If there is not a planned outage then try to restart the Nimbus on this server:

```
root# nimbus restart
```

4. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Messages from CallAgent Server

Maintenance Due to CallAgent Message

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> msgs

Description:
The CallAgent treats the message exchange with the MGCP MTA. The CallAgent has an all active redundancy scheme. If one CallAgent fails the remaining CallAgent take over the work load.

Consequences:

Warning	This erroneous condition must be checked within short time!

For the VoIP Switch telephony service:

◊ As long one CallAgent remains the VOIP Switch works

For the operations:

◊ None

For the user:

◊ Single MGCP MTA at the user's premises is not working correctly. The telephone service may not always work for this users.

Solution:
Depends on the analyzed problem.

Action:

1. Check if the IP network is OK

2. Check the status of the CallAgent components

Confirm that the reported CallAgent server is affected

3. Check the reported CallAgent server with the "Server Administrator (OMSA)"

4. Treat the problem:

a) If there are IP network problems

Actions see: "Maintenance Due to IP Network Alarm"

b) If there is a CallAgent problem try to restart the component:

```
root# callagent restart
```

5. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Messages from CPECenter Server

Maintenance Due to CpeCenterMessage

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> msgs
<HOST_NAME> msgs "DevAdmProvider (-1) duplicated devicetype:" <DEVICE_TYPE>

Description:
During the preparation of a device configuration file two device configuration templates were found. If a CPE loads a device configuration file which was produced under these conditions it may not work correctly.

Consequences:

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ None

For the user:

◊ The CPE may not work with the produced configuration file

Solution:
One device configuration template has to be deleted.

Action:
1. Contact the "VoIP Switch Supplier Support"!

Maintenance Due to IP Network Alarms

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> conn "Host does not respond to ping" <IP_ADDRESS>

Description:
This test performs a "ping" toward the IP address of the host. If the "ping" is not answered then there is a problem with the IP network, e.g.:

◊ Pinged host defect or off line
◊ Layer2 IP Switch defect or off line
◊ Brocken IP backbone network

Consequences:

Warning	**MOST SEVERE** condition if several VoIP Switch server are affected for a longer duration (ca 15min)!

For the VoIP Switch telephony service:

◊ The telephone service may be interrupted

For the operations:

◊ The MySQL databases may loose their replication

For the user:

◊ The telephone service may be interrupted for the users!

Solution:
Solve the IP network problems!

Check the IP network devices:

◊ Pinged host
◊ Layer 2 IP switches
◊ IP Routes
◊ Firewalls

Check the VoIP Switch server IP connectivity.

Action:
1. Evaluate the severity of the IP network outage:

a) Check if it is a occasional ping failure:

- Only one host doesn't respond
- Only 1 or 2 poll cycle fail
  Type "Occasional Failure":
- In this situation the erroneous situation may be neglected.

b) Check if it is only a single host:

- One host doesn't respond anymore
  Type "Host Failure":
  ◊ Check the hardware condition and IP connectivity of this device
  ◊ Check with the VoIP Switch Administrator in the ConfigCenter Menu "Components" how the VoIP Switch is affected

c) Check if more than one VoIP Switch server is affected:

- More than one VoIP Switch server don't respond anymore
  Type "VoIP Switch Failure":

1. Check with the VoIP Switch Administrator how the VoIP Switch is affected:

   a) Connect to both (*-ms-01, *-ms-02) ConfigCenter Menu "Components" and check the component status

   b) Check the questions:

   ◊ Which VoIP Switch servers are not visible?
   ◊ Are they the same on both ConfigCenter?

   ◊ Does one ConfigCenter see only the servers on its side? E.g.:
   Side A components complain that they doesn't see their peers on Side B?
   Side B components complain that they doesn't see their peers on Side A?
   If yes => There is a heavy IP backbone problem

   c) Check in the ConfigCenter Menu **Channles** if new connections were established since the IP outage

   If yes => Some users still can make phone calls

2. Check with the VoIP Switch Administrator how the users are affected:

   a) Connect to both (*-ms-01, *-ms-02) Xymon Column "regs" and check the CPE and MTA registrations status.

   b) Check the questions:

   ◊ Check: Do drop the user's CPE registration?
   If yes => There is a heavy IP backbone problem some users cannot use the telephony service anymore!

3. Treat the Type "VoIP Switch Failure":

   a) VoIP Switch Administrator:

   In this situation the erroneous situation may be neglected. Observe if the situation remains.

2. Treat the Type " Occasional Failure ":

   a) VoIP Switch Administrator:

   If possible pre-bar the VoIP Switch component on this server

   b) Solve the IP or hardware issue with the failed host

3. Treat the Type "VoIP Switch Failure":

   a) VoIP Switch Administrator:

   Contact the "VoIP Switch Supplier Support"

4. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Operating System Alarms

The VoIP Switch Administrator and/or server service personnel find here instructions for managing problems indicated by the operating system supervision.

Maintenance Due to Supervised Processes Missing

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> procs "Processes not OK" <MISSING_PROCESS>
```

Description:
One or more supervised process of a Linux service or VoIP Switch component is missing.

Consequences:

Warning **SEVERE erroneous condition** that must be handled!

For the VoIP Switch telephony service:

- ◊ Depends If a VoIP Switch component is missing then the VoIP Switch looses redundancy capability
- ◊ If a Linux service is missing the VoIP Switch may be hampered or the server is not working correctly

For the operations:

- ◊ Depends on the VoIP Switch components or Linux service

For the user:

- ◊ Depends on the VoIP Switch components or Linux service

Solution:
Restart the VoIP Switch component or Linux service.

Action:
1. Check with the VoIP Switch Administrator if it is possible to restart the component or service without endangering the VoIP Switch telephony service.

If possible pre-bar the VoIP Switch component via the ConfigCenter!

2. Restart the VoIP Switch component or Linux service:

a) Restart the VoIP Switch component

```
root# <COMPONENT> restart
```

◊ Example:

```
root# servicecenter restart
```

b) Restart the service:

```
root# /etc/init.d/<SERVICE> restart
```

◊ Example:

```
root# monit restart
```

3. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Supervised IP Ports

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> ports "Ports not OK" <MISSING_PROCESS_PORTS>
```

Description:
One or more supervised IP port of a Linux service or VoIP Switch component is missing.

Consequences:

Warning **SEVERE erroneous condition** that must be handled!

For the VoIP Switch telephony service:

- ◊ Depends If a VoIP Switch component is missing then the VoIP Switch looses redundancy capability
- ◊ If a Linux service is missing the VoIP Switch may be hampered or the server is not working correctly

For the operations:

- ◊ Depends on the VoIP Switch components or Linux service

For the user:

- ◊ Depends on the VoIP Switch components or Linux service

Solution:
Restart the VoIP Switch component or Linux service.

Action:
1. Check with the VoIP Switch Administrator if it is possible to restart the component or service without endangering the VoIP Switch telephony service.

If possible pre-bar the VoIP Switch component via the ConfigCenter!

2. Restart the VoIP Switch component or Linux service:

a) Restart the VoIP Switch component

```
root# <COMPONENT> restart
```

◊ Example:

```
root# servicecenter restart
```

b) Restart the service:

```
root# /etc/init.d/<SERVICE> restart
```

◊ Example:

```
root# monit restart
```

3. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Supervised Hard-Disk Usage

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> disk "File systems not OK"
```

Description:
A hard-disk or hard-disk partition is full. If a hard-disk is full then the Linux operating system behaves unpredictable and the server will most probably crash.

Consequences:

Warning **SEVERE erroneous condition** that must be handled!

For the VoIP Switch telephony service:

◊ Depends on the VoIP Switch components running on the server

For the operations:

◊ Depends on the VoIP Switch components running on the server

For the user:

◊ Depends on the VoIP Switch components running on the server

Solution:
Identify big files or directories. Delete or archive files externally.

Action:
1. Check hard-disk usage:

```
root# df -h
```

2. Find fat files:

```
root# ls -lahS $(find / -type f -size +100k)
```

◊ Example find file sizes >60MByte:

```
root# ls -lahS $(find /opt/backup/ -type f -size +60000k)
```

◊ Check for fat files in the following suspicious directories:
`/opt/backup/`

◊ Do not touch big files in:
`/var/lib/mysql/`

3. Find big directories:

```
root# du -hs
```

Example of a more specific search find directory sizes >1GByte:

```
root# du -hs /home/ratingcenter/* | grep G
root# du -hs /home/*/* | grep G
```

◊ Check the following suspicious directories:

```
/opt/backup/
/home/mediacenter/messages
//home/ratingcenter/cdrs
```

4. Prior of deleting files or directories check with the VoIP Switch Administrator if they are not needed anymore!

If you are suspicious but not sure if it is wise to delete a certain file or directory then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Supervised Memory Usage

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> memory "Memory low"
```

Description:
One or more processes consume a lot of memory space. If the memory becomes low the operating system Linux start to swap memory to and from hard-disk. This reduces the performance of the server.

Consequences:

Warning	This erroneous condition must be handled within reasonable time!

For the VoIP Switch telephony service:

◊ Depends on the VoIP Switch components running on the server

For the operations:

◊ None

For the user:

◊ None

Solution:
Identify which process or consumes the memory. Restart the process in order to free memory. Stop and restart the swapping on the server.

Action:

1. If a LoadBalancer *-lb-* or ServiceCenter *-sc-* server is affected:

Contact the "VoIP Switch Supplier Support"!

2. Find which processes use the memory:

◊ This is a difficult task!

```
root# top
```

3. Stop and restart the swapping:

Preconditions:

◊ Choose a day time where the server is not in high load.
◊ If possible pre-bar the VoIP Switch components on this server via the ConfigCenter
◊ Make sure that the redundant VoIP Switch component is running

a) Restart the responsible process:

```
root# /etc/init.d/<PROCESS_NAME> restart
```

b) Stop the swapping:

◊ Don't do this during high load!
◊ It will take some time until accomplished!

```
root# swapoff -a
```

c) Restart the swapping:

```
root# swapon -a
```

d) Check if the swap is working regularly:

```
root# swapon -s
```

Maintenance Due to Supervised CPU Load

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> cpu "Load is High"

Description:
One or more processes consume extensively CPU power. This may reduce the performance of the server.

Consequences:

Warning	This erroneous condition must be handled within reasonable time!

For the VoIP Switch telephony service:

◊ Reduced performance on the affected server and VoIP Switch component

For the operations:

◊ None

For the user:

◊ None

Solution:
The CPU consuming process has to be identified. If a process is identified it has to be checked if it is a regular or erroneous situation.

If it is a regular situation then it has to be investigated if the servers computing power is still sufficient for this VoIP Switch. If the server hosts a VoIP Switch component which offers an configurable load acceptance via the ConfigCenter then it is worth a try to reduce the components workload.

An erroneous situation can mostly be solved by restarting the process.

Action:
1. Identify the responsible process:

a) Check the process situation with:

```
root# top
root# ps aux
```

b) If a process is suspicious check for multiple processes of the same name:

```
root# ps -aef
```

c) If a process is suspicious check for zombie processes (lists the zombie process id):

```
root# ps aux
```

d) Evaluate with the VoIP Switch Administrator if the suspicious process is in a regular or erroneous state.

2. Handle an erroneous Linux process state.

a)* Restart a Linux process:

```
root# /etc/init.d/<PROCESS_NAME> restart
```

b) Kill a process, e.g. double started process, zombie:

```
root# kill -9 <PROCESS_ID>
```

3. Handle a VoIP Switch component :

a) Restart an erroneous VoIP Switch component:

```
root# <COMPONENT_NAME> restart
```

b) If the VoIP components ServiceCenter or MediaServer produces high load then the VoIP Switch Administrator may reduce their accepted work load via the ConfigCenter.

4. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

Maintenance Due to Supervised Files Missing or to Big

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
<HOST_NAME> ????

Description:

Consequences:

For the VoIP Switch telephony service:

◊ None

For the operations:

◊ None

For the user:

◊ None

Solution:

Action:
1. If the erroneous condition remains then contact the "VoIP Switch Supplier Support"!

VoIP System Maintenance

Best Practice for Handling a "Fraud" Situation

The Aarenet VoIP Switch Administrator finds here instructions for managing fraud problems.

1. Immediate action:

◊ Block call routing to the destination (usually somewhere in the Caribbean, west or central Africa)
◊ If only from one source IP address then block this IP address on the FW

2. Investigate if the fraud is due to "Direct Registrations" with correct SIP credentials on the VoIP Switch:

◊ Check if the calling number has multiple SIP registrations of a suspicious source IP range or user agent!
If YES then:
The SIP credentials were not kept secret or hacked from the users CPE

Action:

- Block this user account for outgoing calls (blocking international calls is usually sufficient)
- Change the SIP credential in the user account and the user's CPE.
- Change the CPE administration login credentials

3. Investigate if the fraud is due to "Hacked Users CPE":

a) Analyze the traces of some fraud connections.

Check if the source IP remain the one of a registered user CPE!
If YES then:
If yes block this user account for outgoing calls

Action:

- Block this user account for outgoing calls (blocking international calls is usually sufficient)
- Inform the user about the fraud and its reason
- Change the SIP credential in the user account and the user's CPE.
- Change the CPE administration login credentials

4. Post Work:

◊ Undo the "immediate action"
◊ Enable the customer account when the SIP credentials and CPE administration login credentials are changed

-->

Guide for the Maintenance and Problem Solving for Servers from DELL Inc ®

Best Practice When a Hardware HW Problem is Indicated

It is assumed that from any source a hardware problem of a server is indicated, e.g.:

◊ Monitor Log
◊ Alerting email
◊ SMTP trap
◊ system engineer observation
◊ etc

Best Practice	1. Access the server's "OpenManage Server Administrator (OMSA)" GUI → Show me how ... 2. Check the server's hardware problem → Show me how ... 3. Prepare documentation for a ticket at the DELL support: · Description of the problem · Get the servers Service Tag → Show me how ... · Get the server log → Show me how ... 4. Organize the hardware part replacement if needed → Show me how ... 5. Treat the hardware problem: · Replace the defect hardware part → Show me how ... · Replace the defect hard-disk → Show me how ... · Restart the RAID replication → Show me how ...

Server Monitoring

Manual Server Monitoring With DELL's "Server Administrator (OMSA)"

DELL OpenManage Server Administrator (OMSA) is a software agent that provides a comprehensive, one-to-one systems management solution in two ways: from an integrated, Web browser-based graphical user interface (GUI) and from a command line interface (CLI) through the operating system.

Note	In this chapter enough information is given for being dangerous! If there are uncertainties contact the "DELL Support" or the "VoIP Switch Supplier Support".

Access the "OpenManage Server Administrator (OMSA)"

Connect with any Web browser to the server's "OpenManage Server Administrator (OMSA)" GUI:

1. Insert the following URI:
   `https://<IP_ADDRESS>:1311`
   Example:
   `https://172.100.100.100:1311`
2. Insert the user "root" login credentials:
   - Username: `root`
   - Password: *the server root password*

Check the Type of Server and Service Tags

Access the server's "OpenManage Server Administrator (OMSA)" GUI.

Check the server type:

◊ In the OMSA home page menu bar at the top the server type is listed, e.g.: "PowerEdge620"

or

◊ Menu "System" Tab "Properties" Tab "Summary"

Check the Service Tag:

◊ Menu "System" Tab "Properties" Tab "Summary"
In frame "Main System Chassis" the Service Tag is displayed, e.g. : `47X....`
In frame "Main System Chassis" the "Express Service Code" is displayed, e.g. : `9187....`

Check the Server's Hardware Status

Access the server's "OpenManage Server Administrator (OMSA)" GUI.

Check the Server's Hardware Status:

◊ Menu "System" Tab "Properties" Tab "Health"
◊ Click "Main System Chassis"
The status of all server hardware components is displayed and can be checked in detail.

Check the Server's and RAID and Hard-Disk HD Status

Access the server's "OpenManage Server Administrator (OMSA)" GUI.

Check the RAID Controller Type:

◊ Menu "System" Tab "Properties" Tab "Health"
◊ Click "Storage"
In frame "RAID Controller(s)" the RAID controller type is displayed, e.g. : `"PERC 6/i integrated"`

Check the RAID Controller Status:

◊ Menu "System" Tab "Properties" Tab "Health"
◊ Click "Storage"
In frame "RAID Controller(s)" the name and status of the RAID is displayed: `"Virtual Disk 0 RAID-1"`

Check the Hard-Disk HD Replication Status

Access the server's "OpenManage Server Administrator (OMSA)" GUI.

Check the Hard-Disk HD Status:
You have to dig in via the left navigation tree:

◊ Menu "Storage" Menu "PERC ..." Menu "Connector ..." Menu "Enclosure ..." Menu "Physical Disks ..."
Check the disk state: `Column "State"`

States:

◊ Online:
- The disk is online and productive working in the RAID. The replication is working.

◊ Ready:
- The disk is ready for integration into a RAID. The replication is not active.

◊ Rebuilding:
- The disc is currently integrated into the RAID. The progress is displayed in %.

If there is an indication of a hard-disk replication problematic then check in chapter "Treating RAID and Hard-Disk Problems" about further maintenance actions.

Get the Server's Log Data

Access the server's "OpenManage Server Administrator (OMSA)" GUI.

Get the OMSA log:

◊ Menu "System" Tab "Logs"
◊ Save the "Embedded System Management (ESM) Log" on the server:
Click "Save AS" and follow the instructions
◊ Copy the saved EMS Log file to the support directory of the case

Server Monitoring by Xymon

The VoIP Switch default monitor Xymon is described in "VoIP Switch Monitoring"

Indication of a Server Hardware Defect

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> "snmptrapd" "failure"`

Description:

The server indicates any hardware failure:

◊ Failed power module
◊ Failed main board
◊ Failed RAID controller
◊ Failed hard-disk
◊ Any other hardware problem

Consequences:

Warning	It may be a **SEVERE** server condition that must be immediately investigated and treated!

For the VoIP Switch telephony service:

◊ Depends on the VoIP Switch components running on the server

For the operations:

◊ Depends on the VoIP Switch components running on the server

For the user:

◊ Depends on the VoIP Switch components running on the server

Solution:
The server must be repaired or exchanged.

Action:

1. Check the details on the server with the "Server Administrator (OMSA)"
2. Organize DELL repair parts according the maintenance agreement with your "VoIP Switch Supplier"
   - Direct at DELL support
   - Contact the "VoIP Switch Supplier Support"
3. Repair the server:
   - Default processing of hardware problems that forces to shutdown the server, e.g.:
     - Fix main board
     - Fix RAID controller
     - Fix or wear out batteries
     - Fix fan
     - Fix RAM modules

   or
   - Processing of hardware problems that can be done hot, e.g.:
     - Fix power module
     - Fix hard-disk

Indication of a Server Hard-Disk or RAID Controller Problem

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> "snmptrapd" "degraded"`

Description:
The server indicates a problem with the virtual disk:

◊ Failed RAID controller
◊ Failed hard-disk
◊ Failed hard-disk replication

Consequences:

Warning	**SEVERE** server condition that must be immediately investigated and treated!

For the VoIP Switch telephony service:

◊ Depends on the VoIP Switch components running on the server

For the operations:

◊ Depends on the VoIP Switch components running on the server

For the user:

◊ Depends on the VoIP Switch components running on the server

Solution:
The RAID controller must be repaired or a hard-disk exchanged.

Action:

1. Check the details on the server with the "Server Administrator (OMSA)"
2. Organize DELL repair parts according the maintenance agreement with your "VoIP Switch Supplier"
   - Direct at DELL support
   - Contact the "VoIP Switch Supplier Support"
3. Repair the server:
   - Default processing of hardware problems that forces to shutdown the server, e.g.:
     - Fix RAID controller

   or
   - Processing of hardware problems that can be done hot, e.g.:
     - Fix hard-disk
     - Fix hard-disk replication

Procedure for Replacing Defect HW Parts with DELL

The procedure for exchanging defect hardware HW of DELL servers' is different from country to country and may also change from time to time.

The following basic procedure for HW exchange seems more or less stable:

1. Detect the HW problem
2. Make sure to have ready the DELL server details:
   - Server Type
   - Service-Tag number or the "ExpressService Code"
   - Check the guaranty time of the server
3. Report DELL support
   - DELL will analyze the case and order more information if needed
4. DELL will organize and send the exchange part
5. The VoIP Switch Administrator has to organize the replacing of the part
   Usually this has to be done within **1 - 3 working days**
6. The VoIP Switch Administrator has to make ready the defect part for returning it to DELL
   - **Do not dispose the defect part!**
     Either the defect part will be picked up at the location or it has to be send back to DELL.

Treating Server Hardware Problems

The VoIP Switch Administrator and/or server service personnel find here instructions for managing HW defects.

Default Process for Fixing Hardware Problems

Indication:

◊ Xymon Event either email and/or SNMP trap:
- Hardware failed
- RAID degraded
- Host does not respond to ping
- Ports not OK
- Processes not OK

◊ The provider's system monitoring indicates no access to the server

◊ Server Administrator (OMSA): Displays the error condition
◊ Server Display: The server front display is yellow and indicates the error condition
◊ Server Console: The server doesn't respond to console input

Description:
Any hardware problem.
Most probably:

◊ Defect main board
◊ Defect RAID controller
◊ Defect or wear out batteries
◊ Defect fan
◊ Defect power module

Note	The telephony service for the customers is not endangered as long only one server fails! It becomes disastrous if the two LoadBalancer servers or all ServiceCenter servers are not working anymore.

Consequences:

Warning	It may be a **SEVERE** server condition that must be immediately investigated and treated!

For the VoIP Switch telephony service:

◊ Depends on the VoIP Switch components running on the server
◊ If a ServiceCenter server fails the capability of concurrent connection handling may decline.

For the operations:

◊ Depends on the VoIP Switch components running on the server

For the user:

◊ Depends on the VoIP Switch components running on the server

Solution:
The server must be repaired or exchanged.

Action:
Analyze the situation and organize spare parts:

1. Check the details on the server with the "Server Administrator (OMSA)"
2. Organize DELL repair parts according the maintenance agreement with your "VoIP Switch Supplier"
   - Direct at DELL support
   - Contact the "VoIP Switch Supplier Support"

Treat the VoIP Switch operation if the defect stops the proper server functionality :

1. Disable Xymon Alarming
2. Stop provider alarming
3. Graceful pre-bar the VoIP Switch component

Repair the server:
If the main board or RAID controller had to be replaced then follow these special instructions:

◊ Fix main board
◊ Fix RAID controller

If the power-module or hard-disk have to be replaced, see:

◊ Fix power module
◊ Fix hard-disk

Warning	For the following actions the server casing has to be opened! The effects of EMC must be considered and the appropriate precautions must be taken to prevent further hard ware damage.

1. Shut down and power off the server if the part has to be replaced on the main board
2. Repair the server Follow the server manufacturer's instructions!

Put back the server to normal working state:

1. Start the server (if needed):
   This automatically starts the VoIP Switch components!
2. Checks:
   1. Check the server status with "Server Administrator (OMSA)"
   2. Check in the ConfigCenter if all VoIP Switch components on the sever are ok:
      `ConfigCenter GUI  Menu "System"  Menu "Components"`
   3. Check if the Xymon monitor doesn't show any error

If the VoIP Switch doesn't get back to normal telephony service operation:

1. Investigate what is wrong and solve it
2. Contact the "VoIP Switch Supplier Support" for helping setting up the server and recovering the missing VoIP Switch functionality

Enable the alarming again:

1. Enable Xymon Alarming
2. Start provider alarming

Fix Defect Main Board or RAID Controller

See section "Default Process for Fixing Hardware Problems" for the general description of the problem.

Actions:

Repair the server:

1. Shut down and power off the server if the part has to be replaced on the main board
2. Repair the server hardware Follow the server manufacturer's instructions
3. Connect a VGA monitor to the console port of the server

If the RAID controller was repaired then there will be still a RAID problem continue at "Default Process for Fixing RAID Problems", Case 2

If the main board was repaired continue here:

1. Insert the original hard-disk 1 in bay 0 (do not insert the hard-disk 2 yet)

Put back the server to normal working state:

1. Power on and start the server
   This automatically starts the VoIP Switch components!
2. Checks:
   1. Check the console output on the VGA monitor if any exceptions are displayed during the BIOS booting
      If the booting stucks during virtual hard disk initialization (RAID controller) then check the replication issues .
   2. Check the server status with "Server Administrator (OMSA)"
   3. Check in the ConfigCenter if all VoIP Switch components on the sever are ok:
      `ConfigCenter GUI  Menu "System"  Menu "Components"`
   4. Check if the Xymon monitor doesn't show any error:
      After a certain time all supervised objects should get green except the missing hard-disk 2

If the VoIP Switch doesn't get back to normal telephony service operation:

1. Investigate what is wrong and solve it
2. Contact the "VoIP Switch Supplier Support" for helping setting up the server and recovering the missing VoIP Switch functionality

When the server and the telephony service are working correctly again then:

1. Insert the original hard-disk 2 in bay 1
   - Check with "Server Administrator (OMSA)" if the RAID controller started automatically the hard disk replication if not then restart the replication manually

Enable the alarming again:

1. Enable Xymon Alarming
2. Start provider alarming

Fix Defect Power Module

Indication:

◊ Xymon Event either email and/or SNMP trap:
- Hardware failed

◊ Server Administrator (OMSA): Displays the error condition
◊ Server Display: The server front display is yellow and indicates the error condition

Description:
Defect power module

Consequences:

Note	This erroneous condition must be checked and treated within reasonable time!

For the VoIP Switch telephony service:

◊ No immediate consequences
◊ The server is running just with one power module

For the operations:

◊ No immediate consequences

For the user:

◊ No immediate consequences

Solution:
The power module must be replaced

Actions:

Analyze the situation and organize spare parts:

1. Check the details on the server with the "Server Administrator (OMSA)"
2. Organize DELL repair parts according the maintenance agreement with your "VoIP Switch Supplier"
   - Direct at DELL support
   - Contact the "VoIP Switch Supplier Support"

Treat the VoIP Switch operation if the defect stops the proper server functionality :

1. Disable Xymon Alarming
2. Stop provider alarming

Replace the power module:

1. Remove the defect power module (hot plug out possible)
2. Insert the new power module (hot plug in possible)
3. Connect the power cord

Put back the server to normal working state:

1. Checks:
   1. Check the server status with "Server Administrator (OMSA)"
   2. Check if the Xymon monitor doesn't show any error

If the server doesn't go back to normal operation:

1. Investigate what is wrong and solve it
2. Contact the "VoIP Switch Supplier Support" for helping recovering the server

Enable the alarming again:

1. Enable Xymon Alarming
2. Start provider alarming

Treating RAID and Hard-Disk Problems

All servers of the VoIP Switch run a RAID type 1 which mirrors the contents of the two installed hard-disks. The "RAID controller" manages the replication between the two hard-disks.

Several conditions may interrupt the hard-disk replication and/or degrade the RAID virtual disk:

◊ Main board defect
◊ RAID controller defect
◊ Hard-disk defect

The consequences are that the server is not running at all or only with one hard-disk. The good news is as long one hard-disk is running the server will work as expected.

Note	These types of defect have to be solved as fast as possible!

Fix Defect Hard Disk

Indication:

◊ Xymon Event either email and/or SNMP trap:
- Hardware failed
- RAID degraded

◊ Server Administrator (OMSA): Displays the error condition
◊ Server Display: The server front display is yellow and indicates the error condition

Description:
Defect hard-disk

Consequences:

Note	This erroneous condition must be checked and treated within reasonable time!

For the VoIP Switch telephony service:

◊ No immediate consequences
◊ The server is running just with one hard-disk

For the operations:

◊ No immediate consequences

For the user:

◊ No immediate consequences

Solution:
The hard-disk must be replaced

Actions:

Analyze the situation and organize spare parts:

1. Check the details on the server with the "Server Administrator (OMSA)"
2. Organize DELL repair parts according the maintenance agreement with your "VoIP Switch Supplier"
   · Direct at DELL support
   · Contact the "VoIP Switch Supplier Support"

Treat the VoIP Switch operation if the defect stops the proper server functionality :

1. Disable Xymon Alarming
2. Stop provider alarming

Replace the hard-disk:

1. Remove the defect hard-disk (hot plug out possible)
2. Insert the new hard-disk (hot plug in possible):
   If the hard-disk is brand-new the replication starts immediately
   If the hard-disk was already used then the replication may not start automatically then check the instructions at " Default Process for Fixing RAID Problems", Case 1 .

Put back the server to normal working state:

1. Checks:
    1. Check if the hard-disk replication is in progress
    2. Check the server status with "Server Administrator (OMSA)"
    3. Check if the Xymon monitor doesn't show any error

If the server doesn't go back to normal operation:

1. Investigate what is wrong and solve it
2. Contact the "VoIP Switch Supplier Support" for helping setting up the hard-disk replication

Enable the alarming again:

1. Enable Xymon Alarming
2. Start provider alarming

Default Process for Fixing RAID Problems

Indication:

◊ Xymon Event either email and/or SNMP trap:
- Hardware failed
- RAID degraded

◊ The provider's system monitoring may indicate no access to the server

◊ Server Administrator (OMSA): Displays the error condition
◊ Server Display: The server front display is yellow and indicates the error condition
◊ Server Console: The server may not respond to console input

Description:
Any hardware problem.
Most probably:

◊ Defect RAID controller
◊ Defect hard-disk

Consequences:

Warning	It may be a **SEVERE** server condition that must be immediately investigated and treated!

For the VoIP Switch telephony service:

◊ Depends on the VoIP Switch components running on the server
◊ If a ServiceCenter server fails the capability of concurrent connection handling may decline.

For the operations:

◊ Depends on the VoIP Switch components running on the server

For the user:

◊ Depends on the VoIP Switch components running on the server

Solution:
The server must be repaired or exchanged.

Action:

A) Analyze the degrade situation and organize spare parts:

1. Check the details on the server with the "Server Administrator (OMSA)"
2. Check the VoIP Switch documentation for the server type and used RAID controller
3. Organize DELL repair parts according the maintenance agreement with your "VoIP Switch Supplier"
   - Direct at DELL support
   - Contact the "VoIP Switch Supplier Support"

B) Treat the VoIP Switch operation if the defect stops the proper server functionality :

1. Disable Xymon Alarming
2. Stop provider alarming
3. :support_switch#supportSwitchPreBar Graceful pre-bar the VoIP Switch component

C) Evaluate the repair case for DELL RAID controller type: PERC5 / PERC 6 / H310 Mini / H320 Mini / H330 Mini:

Case 1: "One Hard-Disk Defect"

Precondition:

- Main board is ok
- RAID controller is ok
- 1 operative hard-disk is ok
- Server is still operative within the VoIP Switch
- The replacement hard-disk has the same form factor and size of bytes

To-Do:

1. Remove the defect hard-disk (hot plug-out is no problem)
2. Insert the new hard-disk (hot plug-in is no problem) either:
   - ♦ a brand-new hard-disk
   - ♦ an already used spare hard-disk
3. Check the hard-disk replication status
   If the replication did not start automatically then start the replication manually !

Case 2: "Main Board or RAID Controller Defect:

Precondition:

- The main board RAID controller are repaired according description above
- 2 operative hard-disks are ok
- Server is shut down
- Disconnect all Ethernet patch cables from the server GB ports.
- Connect a VGA monitor and USB keyboard and mouse tot the console port of the server

To-Do:

1. Insert the original hard-disk 1 in bay 0 (do not insert the hard-disk 2 yet)
2. Power up the server
3. Check the console output on the VGA monitor:
   During the BIOS startup the following message may be displayed:
   ```
   Foreign configuration(n) found on adapter.
   Press any key ? or 'F' to import foreign configuration and
   continue.
   ```
4. If requested press key **F** on the keyboard!
   Note:
   *If you miss to press **F** then restart the BIOS booting by pressing the keys [Ctrl Alt Delete] else the server booting stops after the BIOS start up.*
5. Check the console output on the VGA monitor:
   A security question may be displayed which enables you to stop the procedure:
   ```
   All of the disk from your previous configuration are gone. If this is an
   unexpected message ...
   ```
6. Do not press any key!
   Note:
   If no key is pressed then the RAID controller takes over the hard-disk as part of its new virtual disk.

   Wait until the server has booted!

7. Insert the original hard-disk 2 in bay 1

8. Check the hard-disk replication status
   Note:
   It is very probable that the replication did not start automatically!
   Then:
   At Menu "Storage" a yellow warning triangle is displayed
   Upon click on "Storage" the status is displayed:
   `Virtual Disk 0: degraded`
   If the replication did not start automatically then start the replication manually !

For all other cases:

◊ Contact the "VoIP Switch Supplier Support" for helping setting up the server and recovering the missing VoIP Switch functionality

C) Put back the server to normal working state:

1. If needed connect all Ethernet patch cables to the correct server GB ports
2. Checks:
   1. Check the server status with "Server Administrator (OMSA)"
   2. Check in the ConfigCenter if all VoIP Switch components on the sever are ok:
      `ConfigCenter GUI  Menu "System"  Menu "Components"`
   3. Check if the Xymon monitor doesn't show any error

D) If the VoIP Switch doesn't get back to normal telephony service operation:

1. Investigate what is wrong and solve it
2. Contact the "VoIP Switch Supplier Support" for helping setting up the server and recovering the missing VoIP Switch functionality

E) Enable the alarming again:

1. Enable Xymon Alarming
2. Start provider alarming

Manually Restart the Hard-Disk Replication

In this situation the RAID's virtual disk is in state degraded (only one hard-disk is operative, but two are expected). The RAID controller will automatically grab a free "hot spare" hard-disk and associate it with its degraded virtual disk and start the replication.

Restart the hard-disk replication manually:

1. Connect with any Web browser to the server's "Server Administrator (OMSA)" GUI:
   · Login as user "root"

2. From the inserted 2nd hard-disk the foreign RAID configuration has to be deleted:
   `Menu "Storage"  Menu "PERC xxxxx"`
   `Select at [ Available Task ]: "Clear Foreign Configuration"`
   `<tt>  Click button [ Execute ]`
   `<tt>  Confirm the security check click button [ Clear ]`

3. The inserted 2nd hard-disk has to be declared as "hot spare":
   <tt> Menu "Storage" Menu "PERC xxxxx" "Connector 0" Menu "Enclosure (Backplane)" Menu "Physical Disks"
   `Select at [ Available Task ]: "Assign Global Hot Spare"`
   `<tt>  Click button [ Execute ]`

4. Check the virtual disk replication state:
   <tt> Column "State"

If the hard-disk replication is not starting then contact the appropriate DELL Support or the "VoIP Switch Supplier Support".

`k_the_Hard-Disk_HD_Replication_Status_2">Check the Hard-Disk HD Replication Status`

Access the server's "OpenManage Server Administrator (OMSA)" GUI.

Check the Hard-Disk HD Status:
You have to dig in via the left navigation tree:

◊ Menu "Storage" Menu "PERC ..." Menu "Connector ..." Menu "Enclosure ..." Menu "Physical Disks ..."
Check the disk state: `Column "State"`

States:

◊ Online:
- The disk is online and productive working in the RAID. The replication is working.

◊ Ready:
- The disk is ready for integration into a RAID. The replication is not active.

◊ Rebuilding:
- The disc is currently integrated into the RAID. The progress is displayed in %.

If there is an indication of a hard-disk replication problematic then check in chapter "Treating RAID and Hard-Disk Problems" about further maintenance actions.

Get the Server's Log Data

Access the server's "OpenManage Server Administrator (OMSA)" GUI.

Get the OMSA log:

◊ Menu "System" Tab "Logs"
◊ Save the "Embedded System Management (ESM) Log" on the server:
Click "Save AS" and follow the instructions
◊ Copy the saved EMS Log file to the support directory of the case

Server Monitoring by Xymon

The VoIP Switch default monitor Xymon is described in "VoIP Switch Monitoring"

Indication of a Server Hardware Defect

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:

```
<HOST_NAME> "snmptrapd" "failure"
```

Description:
The server indicates any hardware failure:

◊ Failed power module
◊ Failed main board
◊ Failed RAID controller
◊ Failed hard-disk
◊ Any other hardware problem

Consequences:

Warning	It may be a **SEVERE** server condition that must be immediately investigated and treated!

For the VoIP Switch telephony service:

◊ Depends on the VoIP Switch components running on the server

For the operations:

◊ Depends on the VoIP Switch components running on the server

For the user:

◊ Depends on the VoIP Switch components running on the server

Solution:
The server must be repaired or exchanged.

Action:

1. Check the details on the server with the "Server Administrator (OMSA)"
2. Organize DELL repair parts according the maintenance agreement with your "VoIP Switch Supplier"
   - Direct at DELL support
   - Contact the "VoIP Switch Supplier Support"
3. Repair the server:
   - Default processing of hardware problems that forces to shutdown the server, e.g.:
     - Fix main board
     - Fix RAID controller
     - Fix or wear out batteries
     - Fix fan
     - Fix RAM modules

   or
   - Processing of hardware problems that can be done hot, e.g.:
     - Fix power module
     - Fix hard-disk

Indication of a Server Hard-Disk or RAID Controller Problem

Indication "Xymon Event":
Monitor Log, Email or SMTP Trap may contain the following information:

Indication:
`<HOST_NAME> "snmptrapd" "degraded"`

Description:
The server indicates a problem with the virtual disk:

◊ Failed RAID controller
◊ Failed hard-disk
◊ Failed hard-disk replication

Consequences:

Warning	**SEVERE** server condition that must be immediately investigated and treated!

For the VoIP Switch telephony service:

◊ Depends on the VoIP Switch components running on the server

For the operations:

◊ Depends on the VoIP Switch components running on the server

For the user:

◊ Depends on the VoIP Switch components running on the server

Solution:
The RAID controller must be repaired or a hard-disk exchanged.

Action:

1. Check the details on the server with the "Server Administrator (OMSA)"
2. Organize DELL repair parts according the maintenance agreement with your "VoIP Switch Supplier"
   - Direct at DELL support
   - Contact the "VoIP Switch Supplier Support"
3. Repair the server:
   - Default processing of hardware problems that forces to shutdown the server, e.g.:
     - Fix RAID controller

   or
   - Processing of hardware problems that can be done hot, e.g.:
     - Fix hard-disk
     - Fix hard-disk replication

Procedure for Replacing Defect HW Parts with DELL

The procedure for exchanging defect hardware HW of DELL servers' is different from country to country and may also change from time to time.

The following basic procedure for HW exchange seems more or less stable:

1. Detect the HW problem
2. Make sure to have ready the DELL server details:
   - Server Type
   - Service-Tag number or the "ExpressService Code"
   - Check the guaranty time of the server
3. Report DELL support
   - DELL will analyze the case and order more information if needed
4. DELL will organize and send the exchange part
5. The VoIP Switch Administrator has to organize the replacing of the part
   Usually this has to be done within **1 - 3 working days**
6. The VoIP Switch Administrator has to make ready the defect part for returning it to DELL
   - **Do not dispose the defect part!**
     Either the defect part will be picked up at the location or it has to be send back to DELL.

Treating Server Hardware Problems

The VoIP Switch Administrator and/or server service personnel find here instructions for managing HW defects.

Default Process for Fixing Hardware Problems

Indication:

◊ Xymon Event either email and/or SNMP trap:
- Hardware failed

- RAID degraded
- Host does not respond to ping
- Ports not OK
- Processes not OK

◊ The provider's system monitoring indicates no access to the server

◊ Server Administrator (OMSA): Displays the error condition
◊ Server Display: The server front display is yellow and indicates the error condition
◊ Server Console: The server doesn't respond to console input

Description:
Any hardware problem.
Most probably:

◊ Defect main board
◊ Defect RAID controller
◊ Defect or wear out batteries
◊ Defect fan
◊ Defect power module

Note	The telephony service for the customers is not endangered as long only one server fails! It becomes disastrous if the two LoadBalancer servers or all ServiceCenter servers are not working anymore.

Consequences:

Warning	It may be a **SEVERE** server condition that must be immediately investigated and treated!

For the VoIP Switch telephony service:

◊ Depends on the VoIP Switch components running on the server
◊ If a ServiceCenter server fails the capability of concurrent connection handling may decline.

For the operations:

◊ Depends on the VoIP Switch components running on the server

For the user:

◊ Depends on the VoIP Switch components running on the server

Solution:
The server must be repaired or exchanged.

Action:
Analyze the situation and organize spare parts:

1. Check the details on the server with the "Server Administrator (OMSA)"
2. Organize DELL repair parts according the maintenance agreement with your "VoIP Switch Supplier"
   - Direct at DELL support
   - Contact the "VoIP Switch Supplier Support"

Treat the VoIP Switch operation if the defect stops the proper server functionality :

1. Disable Xymon Alarming
2. Stop provider alarming
3. Graceful pre-bar the VoIP Switch component

Repair the server:
If the main board or RAID controller had to be replaced then follow these special instructions:

◊ Fix main board

◊ Fix RAID controller

If the power-module or hard-disk have to be replaced, see:

◊ Fix power module
◊ Fix hard-disk

Warning	For the following actions the server casing has to be opened! The effects of EMC must be considered and the appropriate precautions must be taken to prevent further hard ware damage.

1. Shut down and power off the server if the part has to be replaced on the main board
2. Repair the server Follow the server manufacturer's instructions!

Put back the server to normal working state:

1. Start the server (if needed):
   This automatically starts the VoIP Switch components!
2. Checks:
   1. Check the server status with "Server Administrator (OMSA)"
   2. Check in the ConfigCenter if all VoIP Switch components on the sever are ok:
      `ConfigCenter GUI  Menu "System"  Menu "Components"`
   3. Check if the Xymon monitor doesn't show any error

If the VoIP Switch doesn't get back to normal telephony service operation:

1. Investigate what is wrong and solve it
2. Contact the "VoIP Switch Supplier Support" for helping setting up the server and recovering the missing VoIP Switch functionality

Enable the alarming again:

1. Enable Xymon Alarming
2. Start provider alarming

Fix Defect Main Board or RAID Controller

See section "Default Process for Fixing Hardware Problems" for the general description of the problem.

Actions:

Repair the server:

1. Shut down and power off the server if the part has to be replaced on the main board
2. Repair the server hardware Follow the server manufacturer's instructions
3. Connect a VGA monitor to the console port of the server

If the RAID controller was repaired then there will be still a RAID problem continue at "Default Process for Fixing RAID Problems", Case 2

If the main board was repaired continue here:

1. Insert the original hard-disk 1 in bay 0 (do not insert the hard-disk 2 yet)

Put back the server to normal working state:

1. Power on and start the server
   This automatically starts the VoIP Switch components!
2. Checks:
   1. Check the console output on the VGA monitor if any exceptions are displayed during the BIOS booting
      If the booting stucks during virtual hard disk initialization (RAID controller) then check the replication issues .
   2. Check the server status with "Server Administrator (OMSA)"
   3. Check in the ConfigCenter if all VoIP Switch components on the sever are ok:
      `ConfigCenter GUI  Menu "System"  Menu "Components"`
   4. Check if the Xymon monitor doesn't show any error:
      After a certain time all supervised objects should get green except the missing hard-disk 2

If the VoIP Switch doesn't get back to normal telephony service operation:

1. Investigate what is wrong and solve it
2. Contact the "VoIP Switch Supplier Support" for helping setting up the server and recovering the missing VoIP Switch functionality

When the server and the telephony service are working correctly again then:

1. Insert the original hard-disk 2 in bay 1
   - ♦ Check with "Server Administrator (OMSA)" if the RAID controller started automatically the hard disk replication if not then restart the replication manually

Enable the alarming again:

1. Enable Xymon Alarming
2. Start provider alarming

Fix Defect Power Module

Indication:

◊ Xymon Event either email and/or SNMP trap:
- Hardware failed

◊ Server Administrator (OMSA): Displays the error condition
◊ Server Display: The server front display is yellow and indicates the error condition

Description:
Defect power module

Consequences:

Note	This erroneous condition must be checked and treated within reasonable time!

For the VoIP Switch telephony service:

◊ No immediate consequences
◊ The server is running just with one power module

For the operations:

◊ No immediate consequences

For the user:

◊ No immediate consequences

Solution:
The power module must be replaced

Actions:

Analyze the situation and organize spare parts:

1. Check the details on the server with the "Server Administrator (OMSA)"
2. Organize DELL repair parts according the maintenance agreement with your "VoIP Switch Supplier"
   - Direct at DELL support
   - Contact the "VoIP Switch Supplier Support"

Treat the VoIP Switch operation if the defect stops the proper server functionality :

1. Disable Xymon Alarming
2. Stop provider alarming

Replace the power module:

1. Remove the defect power module (hot plug out possible)
2. Insert the new power module (hot plug in possible)
3. Connect the power cord

Put back the server to normal working state:

1. Checks:
   1. Check the server status with "Server Administrator (OMSA)"
   2. Check if the Xymon monitor doesn't show any error

If the server doesn't go back to normal operation:

1. Investigate what is wrong and solve it
2. Contact the "VoIP Switch Supplier Support" for helping recovering the server

Enable the alarming again:

1. Enable Xymon Alarming
2. Start provider alarming

Treating RAID and Hard-Disk Problems

All servers of the VoIP Switch run a RAID type 1 which mirrors the contents of the two installed hard-disks. The "RAID controller" manages the replication between the two hard-disks.

Several conditions may interrupt the hard-disk replication and/or degrade the RAID virtual disk:

- ◊ Main board defect
- ◊ RAID controller defect
- ◊ Hard-disk defect

The consequences are that the server is not running at all or only with one hard-disk. The good news is as long one hard-disk is running the server will work as expected.

Note	**These types of defect have to be solved as fast as possible!**

Fix Defect Hard Disk

Indication:

◊ Xymon Event either email and/or SNMP trap:
- Hardware failed
- RAID degraded

◊ Server Administrator (OMSA): Displays the error condition
◊ Server Display: The server front display is yellow and indicates the error condition

Description:
Defect hard-disk

Consequences:

Note	This erroneous condition must be checked and treated within reasonable time!

For the VoIP Switch telephony service:

◊ No immediate consequences
◊ The server is running just with one hard-disk

For the operations:

◊ No immediate consequences

For the user:

◊ No immediate consequences

Solution:
The hard-disk must be replaced

Actions:

Analyze the situation and organize spare parts:

1. Check the details on the server with the "Server Administrator (OMSA)"
2. Organize DELL repair parts according the maintenance agreement with your "VoIP Switch Supplier"
   · Direct at DELL support
   · Contact the "VoIP Switch Supplier Support"

Treat the VoIP Switch operation if the defect stops the proper server functionality :

1. Disable Xymon Alarming
2. Stop provider alarming

Replace the hard-disk:

1. Remove the defect hard-disk (hot plug out possible)
2. Insert the new hard-disk (hot plug in possible):
   If the hard-disk is brand-new the replication starts immediately
   If the hard-disk was already used then the replication may not start automatically then check the instructions at " Default Process for Fixing RAID Problems", Case 1 .

Put back the server to normal working state:

1. Checks:
   1. Check if the hard-disk replication is in progress
   2. Check the server status with "Server Administrator (OMSA)"
   3. Check if the Xymon monitor doesn't show any error

If the server doesn't go back to normal operation:

1. Investigate what is wrong and solve it
2. Contact the "VoIP Switch Supplier Support" for helping setting up the hard-disk replication

Enable the alarming again:

1. Enable Xymon Alarming
2. Start provider alarming

Default Process for Fixing RAID Problems

Indication:

◊ Xymon Event either email and/or SNMP trap:
- Hardware failed
- RAID degraded

◊ The provider's system monitoring may indicate no access to the server

◊ Server Administrator (OMSA): Displays the error condition
◊ Server Display: The server front display is yellow and indicates the error condition
◊ Server Console: The server may not respond to console input

Description:
Any hardware problem.
Most probably:

◊ Defect RAID controller
◊ Defect hard-disk

Consequences:

Warning	It may be a **SEVERE** server condition that must be immediately investigated and treated!

For the VoIP Switch telephony service:

◊ Depends on the VoIP Switch components running on the server
◊ If a ServiceCenter server fails the capability of concurrent connection handling may decline.

For the operations:

◊ Depends on the VoIP Switch components running on the server

For the user:

◊ Depends on the VoIP Switch components running on the server

Solution:
The server must be repaired or exchanged.

Action:

A) Analyze the degrade situation and organize spare parts:

1. Check the details on the server with the "Server Administrator (OMSA)"
2. Check the VoIP Switch documentation for the server type and used RAID controller
3. Organize DELL repair parts according the maintenance agreement with your "VoIP Switch Supplier"
   · Direct at DELL support
   · Contact the "VoIP Switch Supplier Support"

B) Treat the VoIP Switch operation if the defect stops the proper server functionality :

1. Disable Xymon Alarming
2. Stop provider alarming
3. :support_switch#supportSwitchPreBar Graceful pre-bar the VoIP Switch component

C) Evaluate the repair case for DELL RAID controller type: PERC5 / PERC 6 / H310 Mini / H320 Mini / H330 Mini:

Case 1: "One Hard-Disk Defect"

Precondition:

- Main board is ok
- RAID controller is ok
- 1 operative hard-disk is ok
- Server is still operative within the VoIP Switch
- The replacement hard-disk has the same form factor and size of bytes

To-Do:

1. Remove the defect hard-disk (hot plug-out is no problem)
2. Insert the new hard-disk (hot plug-in is no problem) either:
   - ♦ a brand-new hard-disk
   - ♦ an already used spare hard-disk
3. Check the hard-disk replication status
   If the replication did not start automatically then start the replication manually !

Case 2: "Main Board or RAID Controller Defect:

Precondition:

- The main board RAID controller are repaired according description above
- 2 operative hard-disks are ok
- Server is shut down
- Disconnect all Ethernet patch cables from the server GB ports.
- Connect a VGA monitor and USB keyboard and mouse tot the console port of the server

To-Do:

1. Insert the original hard-disk 1 in bay 0 (do not insert the hard-disk 2 yet)
2. Power up the server
3. Check the console output on the VGA monitor:
   During the BIOS startup the following message may be displayed:
   ```
   Foreign configuration(n) found on adapter.
   Press any key ? or 'F' to import foreign configuration and
   continue.
   ```
4. If requested press key **F** on the keyboard!
   Note:
   *If you miss to press **F** then restart the BIOS booting by pressing the keys [Ctrl Alt Delete] else the server booting stops after the BIOS start up.*
5. Check the console output on the VGA monitor:
   A security question may be displayed which enables you to stop the procedure:
   ```
   All of the disk from your previous configuration are gone. If this is an
   unexpected message ...
   ```
6. Do not press any key!
   Note:
   If no key is pressed then the RAID controller takes over the hard-disk as part of its new virtual disk.

   Wait until the server has booted!

7. Insert the original hard-disk 2 in bay 1
8. Check the hard-disk replication status
   Note:
   It is very probable that the replication did not start automatically!
   Then:
   At Menu "Storage" a yellow warning triangle is displayed
   Upon click on "Storage" the status is displayed:
   ```
   Virtual Disk 0: degraded
   ```
   If the replication did not start automatically then start the replication manually !

For all other cases:

- ◊ Contact the "VoIP Switch Supplier Support" for helping setting up the server and recovering the missing VoIP Switch functionality

C) Put back the server to normal working state:

1. If needed connect all Ethernet patch cables to the correct server GB ports
2. Checks:
    1. Check the server status with "Server Administrator (OMSA)"
    2. Check in the ConfigCenter if all VoIP Switch components on the sever are ok:
       `ConfigCenter GUI  Menu "System"  Menu "Components"`
    3. Check if the Xymon monitor doesn't show any error

D) If the VoIP Switch doesn't get back to normal telephony service operation:

1. Investigate what is wrong and solve it
2. Contact the "VoIP Switch Supplier Support" for helping setting up the server and recovering the missing VoIP Switch functionality

E) Enable the alarming again:

1. Enable Xymon Alarming
2. Start provider alarming

Manually Restart the Hard-Disk Replication

In this situation the RAID's virtual disk is in state degraded (only one hard-disk is operative, but two are expected). The RAID controller will automatically grab a free "hot spare" hard-disk and associate it with its degraded virtual disk and start the replication.

Restart the hard-disk replication manually:

1. Connect with any Web browser to the server's "Server Administrator (OMSA)" GUI:
   · Login as user "root"
2. From the inserted 2nd hard-disk the foreign RAID configuration has to be deleted:
   ```
   Menu "Storage"  Menu "PERC xxxxx"
     Select at [ Available Task ]: "Clear Foreign Configuration"
       <tt>  Click button [ Execute ]
         <tt>  Confirm the security check click button [ Clear ]
   ```
3. The inserted 2nd hard-disk has to be declared as "hot spare":
   <tt> Menu "Storage" Menu "PERC xxxxx" "Connector 0" Menu "Enclosure (Backplane)" Menu "Physical Disks"
   ```
   Select at [ Available Task ]: "Assign Global Hot Spare"
     <tt>  Click button [ Execute ]
   ```
4. Check the virtual disk replication state:
   <tt> Column "State"

If the hard-disk replication is not starting then contact the appropriate DELL Support or the "VoIP Switch Supplier Support".

-->

Brief Tutorial of the SIP Signaling and SDP Media Protocols

Knowhow Connection Signaling with "Session Initiation Protocol SIP"

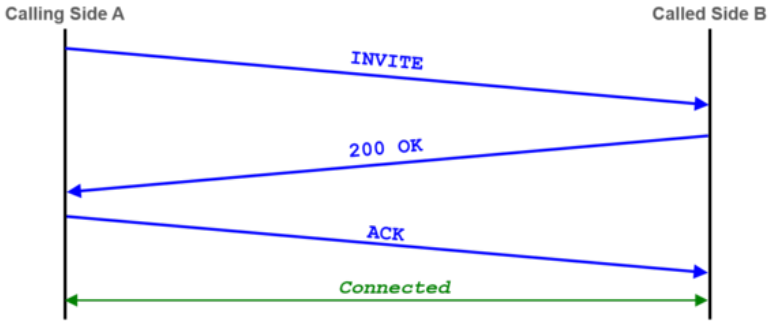
The **Session Initiation Protocol SIP** is a communications protocol for signaling and controlling multimedia communication sessions. One of the most common applications of SIP is in Internet telephony for voice and video calls.

For an extended overview of the SIP protocol visit:

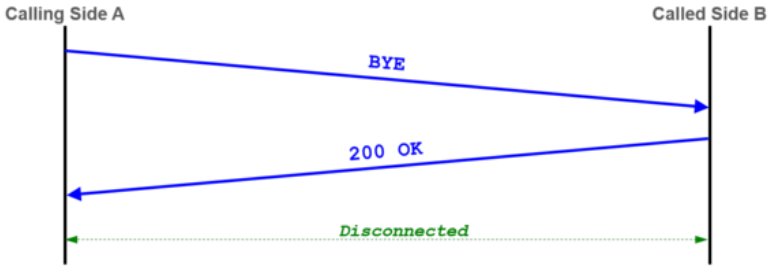
```
Wikipedia: Session Initiation Protocol SIP
```

Basics: Session Session Protocol SIP

Example of a "SIP dialog" with the minimal needed messages for a connection setup or connection renegotiation:

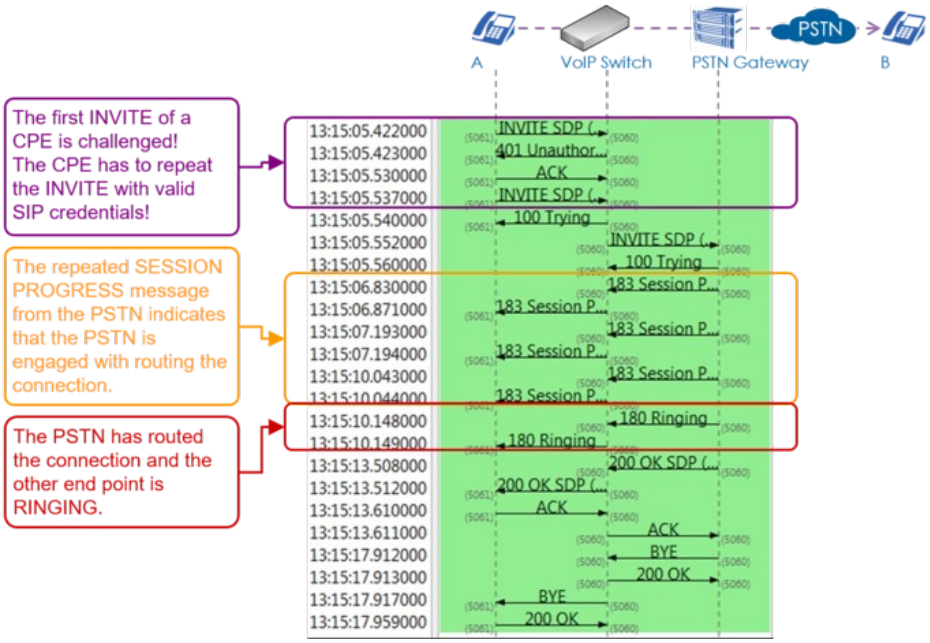


Example of a "SIP dialog" with the minimal needed messages for a connection release:

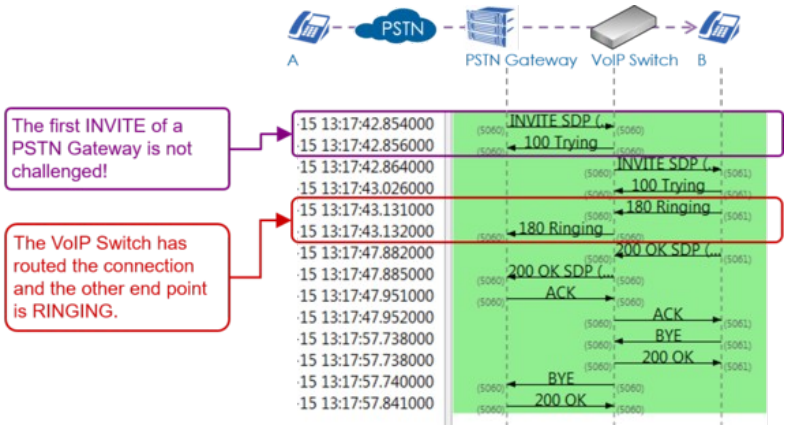


Examples: SIP Signaling Flows

Example of a regular outgoing call into the PSTN:



Example of a regular incoming call from the PSTN:



Example of an outgoing call into the PSTN with three exceptional signaling situations:

1. The PSTN Gateway 1 doesn't respond so the VoIP Switch has to re-route to the PSTN Gateway 2
2. The telephone on side A offers an invalid "Session Time" value which is refused by the PSTN Gateway 2. The telephone on side A has to do a reINVITE with an acceptable "Session Time" value.
3. End point B is busy.

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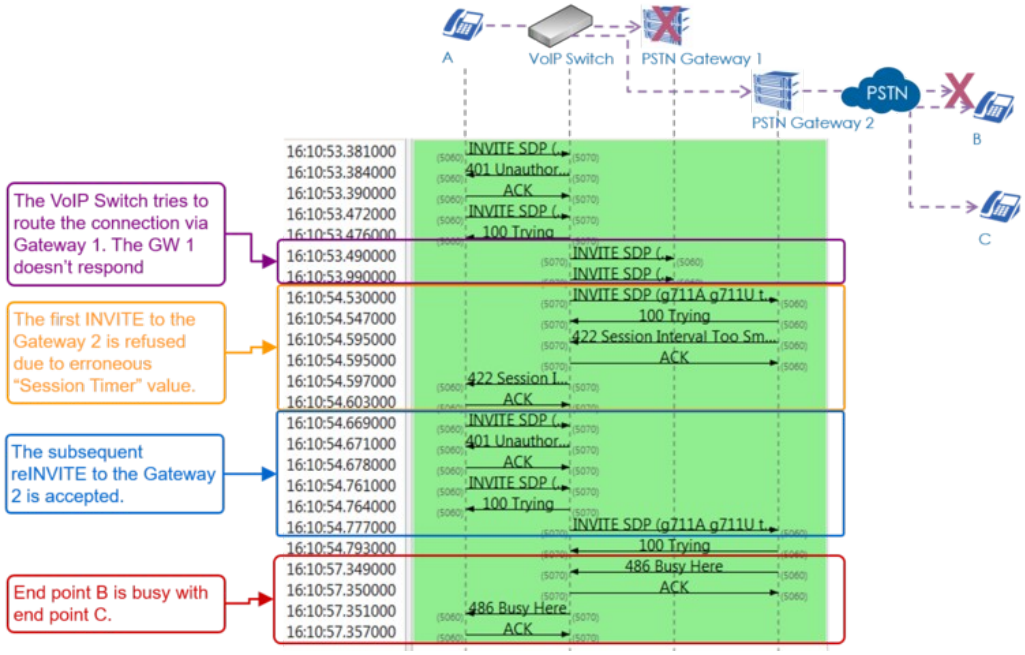
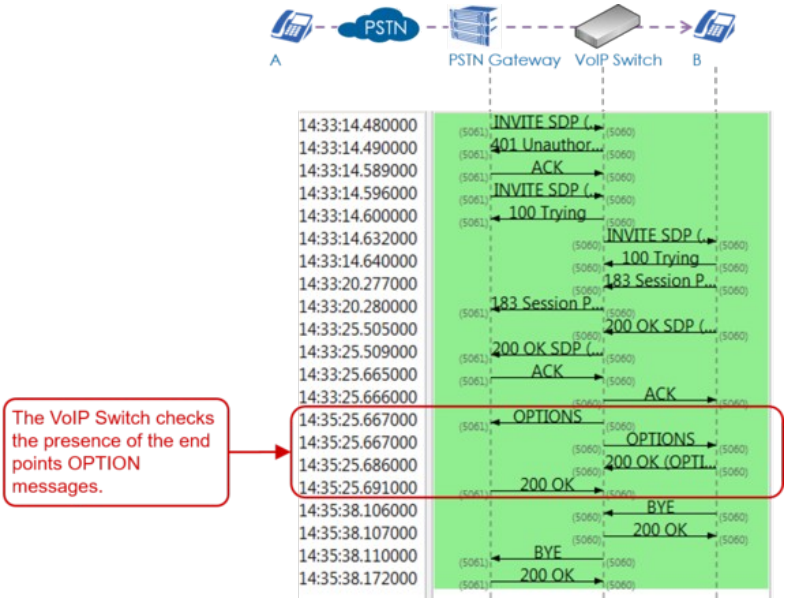


Example of a connection where the VoIP Switch checks the presence of the end points with OPTION messages. The VoIP Switch would release the connection if one end point doesn't respond with "200 OK":



SIP Response Codes

A list of SIP response codes and their meaning can be found here:

```
Wikipedia: List of SIP Response Codes
```

Most Important 1xx?Provisional Responses

100 Trying
Extended search being performed may take a significant time so a forking proxy must send a 100 Trying response.

180 Ringing
Destination user agent received INVITE, and is alerting user of call.

183 Session in Progress
This response may be used to send extra information for a call which is still being set up.

Most Important 2xx?Successful Responses

200 OK
Indicates the request was successful.

Most Important 3xx?Redirection Responses

302 Moved Temporarily
The client should try at the address in the Contact field. If an Expires field is present, the client may cache the result for that period of time.

Most Important 4xx?Client Failure Responses

400 Bad Request
The request could not be understood due to malformed syntax.

401 Unauthorized
The request requires user authentication. This response is issued by UASs and registrars.

403 Forbidden
The server understood the request, but is refusing to fulfil it.

404 Not Found
The server has definitive information that the user does not exist at the domain specified in the Request-URI. This status is also returned if the domain in the Request-URI does not match any of the domains handled by the recipient of the request.

406 Not Acceptable
The resource identified by the request is only capable of generating response entities that have content characteristics but not acceptable according to the Accept header field sent in the request.

408 Request Timeout
Couldn't find the user in time. The server could not produce a response within a suitable amount of time, for example, if it could not determine the location of the user in time. The client MAY repeat the request without modifications at any later time.

410 Gone
The user existed once, but is not available here any more.

480 Temporarily Unavailable
Callee currently unavailable.

486 Busy Here
Callee is busy.

487 Request Terminated
Request has terminated by bye or cancel.

488 Not Acceptable Here
Some aspect of the session description or the Request-URI is not acceptable.

Most Important 5xx?Server Failure Responses

503 Service Unavailable
The server is undergoing maintenance or is temporarily overloaded and so cannot process the request. A "Retry-After" header field may specify when the client may reattempt its request.

Most Important 6xx?Global Failure Responses

603 Decline
The destination does not wish to participate in the call, or cannot do so, and additionally the destination knows there are no alternative destinations (such as a voicemail server) willing to accept the call.

Knowhow Media Stream Signaling with "Session Description Protocol SDP"

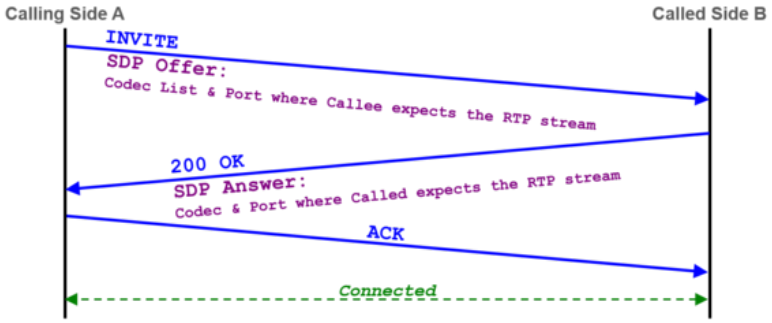
The **Session Description Protocol SDP** describes how during a connection setup the end points negotiate the parameters of this exchange as session announcement, session invitation, and parameter. SDP does not deliver media itself but is used between end points for negotiation of media type, format, and all associated properties for voice, Fax, DTMF, bit transparent data etc..

For an extended overview of the SDP protocol visit Wikipedia.

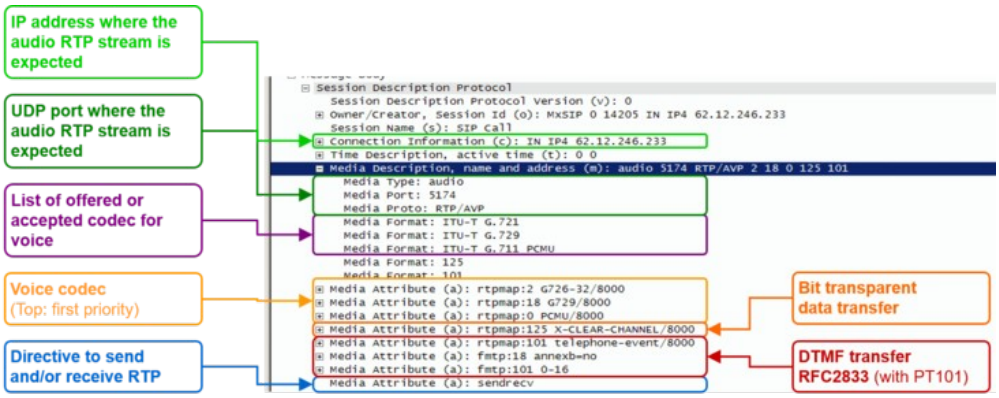
Note	The VoIP Switch doesn't interfere in the SDP negotiation of the end points! There may be exceptions for certain Customer Premises Equipment CPE devices where interoperation problems are known. Check with the VoIP switch administrator which CPE devices are known with SDP manipulations by the VoIP switch.

Basics: Session Description Protocol SDP

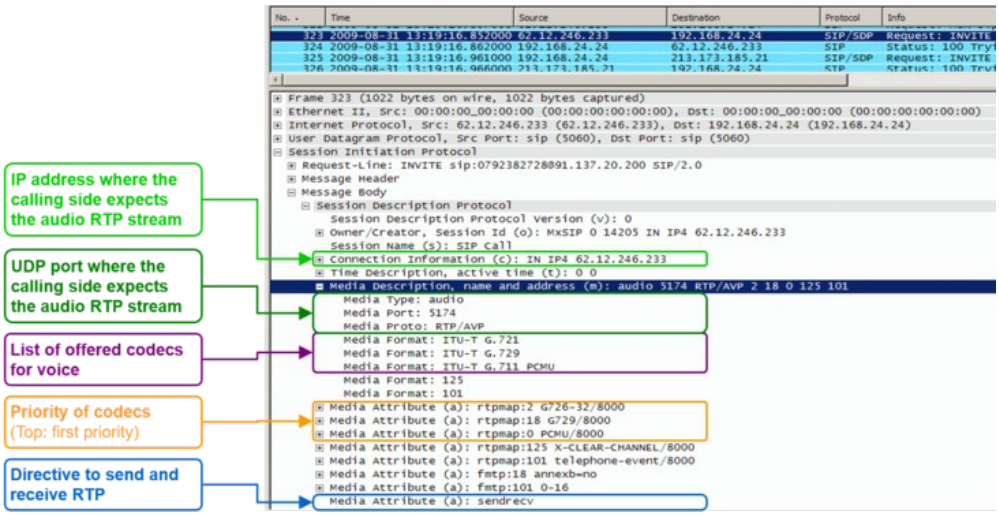
The SDP is embedded in the SIP messages during connection setup or connection renegotiation:



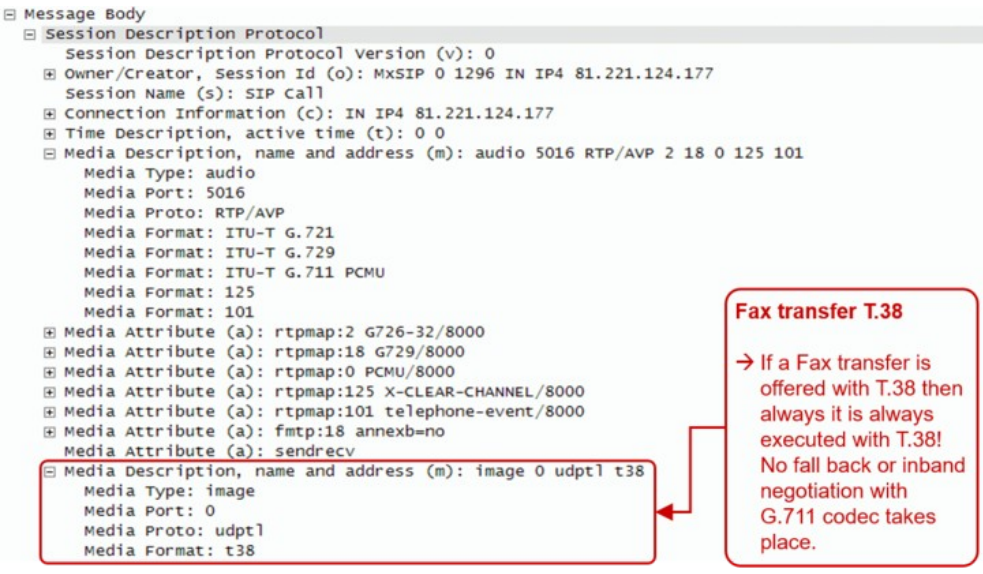
The following SDP properties and parameters are important for supporting customer problems:



Example of a SDP offer from the calling side A:



Example of a SDP offer for a Fax transfer with T.38 from the calling side A:



Interpretation of the "Media Attributes":

Index	Type	Attribute	Remark
0	PCMU	ISDN G.711µlaw	Very good quality VoIP codec
8	PCMA	ISDN G.711alaw	Very good quality VoIP codec
2	G.726-32		Good quality VoIP codec
18	G.729		Low quality VoIP codec

125	x-clear-channel	data service bit transparent	Echo canceling will be switched off and the data bit by bit transferred
101	telephone-event	DTMF, RFC 2833	DTMF will not be transferred inband but as RTP event according RFC 2833
18	annexb=0	Special information for codec with index 18	Special directive for codec G.729
101	0-16	Special information for for telephone-event with index 101	0-15 : DTMF character 0-9, *,#, A,B,C,D 0-16 : DTMF character 0-9, *,#, A,B,C,D, Flash

Basics: RTP/RTCP

The Real Time Protocol RTP is used to transfer media data, e.g. speech in VoIP based telephony.

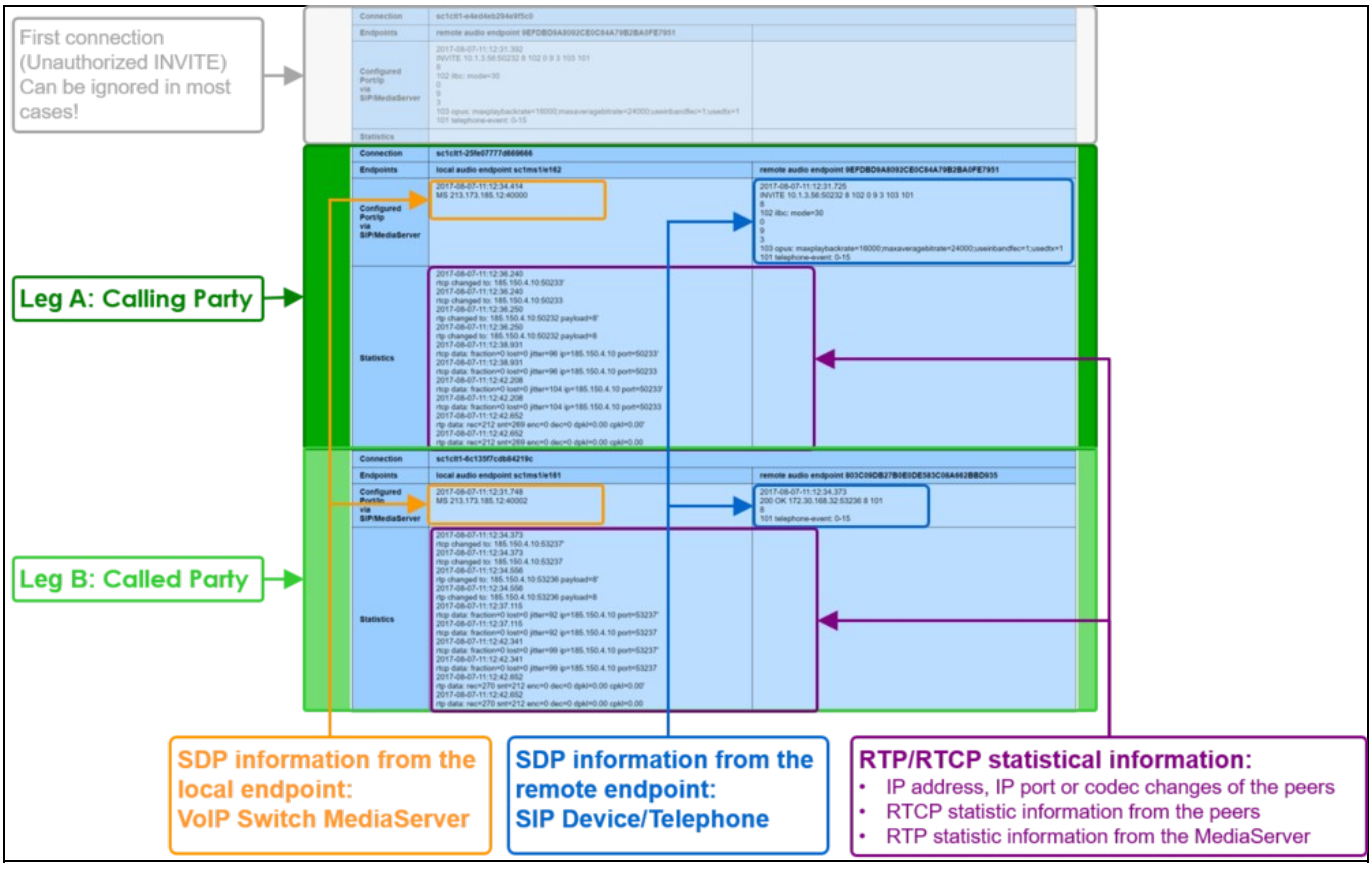
The Real Time Control Protocol RTPC transfers periodically statistical media data between the peers of a connection.

If RTP packets are lost, delayed or jitter then we speak of a Quality of Service QoS problem. For the support it is of interest to know if the number of transferred packets between the peers of a connection and if the numbers in the receive and send paths are reasonable equal, if packets were lost on call leg etc. With these statistical media information it can be possible to identify a path or transfer direction were QoS problems occure.

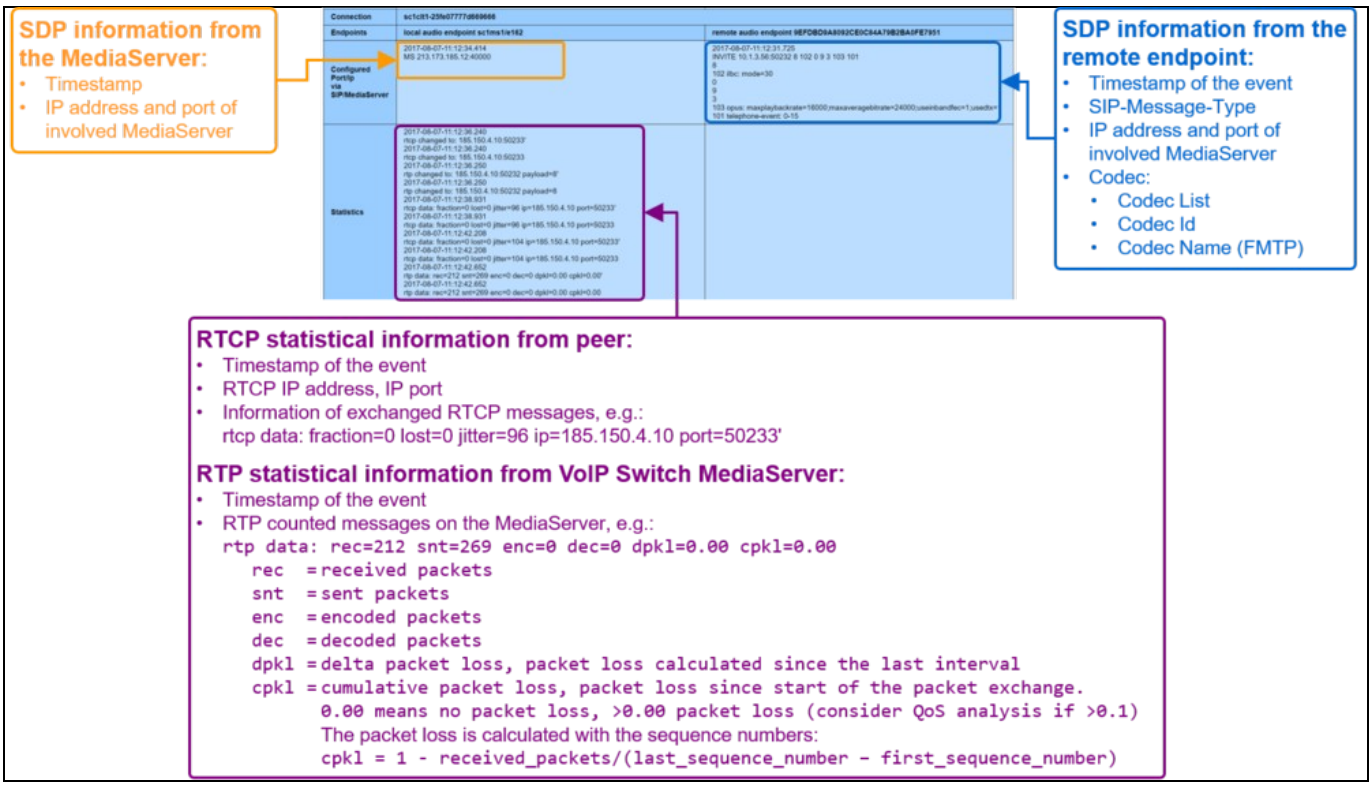
Note	The media stream must be proxied via the MediaServer of the VoIP Switch in order to compute statistical numbers of a connection.

The Aarenet VoIP Switch supports RTP/RTCP statistic data collection of a connection. How they can be obtained is described in article "Manual of the Aarenet VoIP Switch Support Tools", chapter "The ConfigCenter Call Data"

Overview of "RTP/RTCP" information collection:



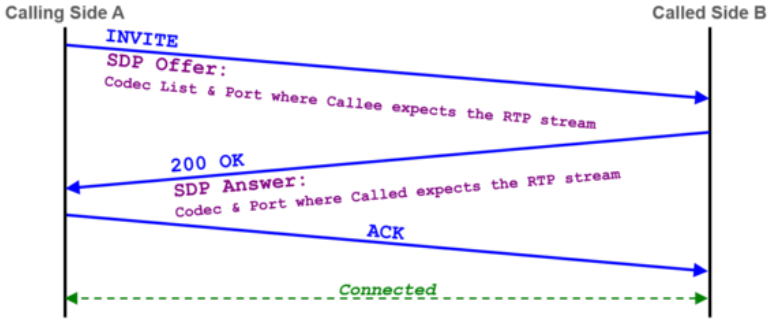
Details of "RTP/RTCP" information collection:



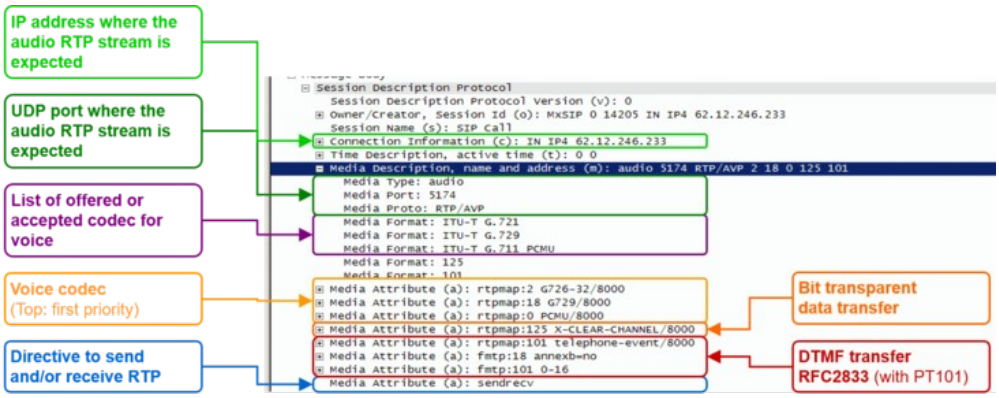
or="#f4cc35">

Basics: Session Description Protocol SDP

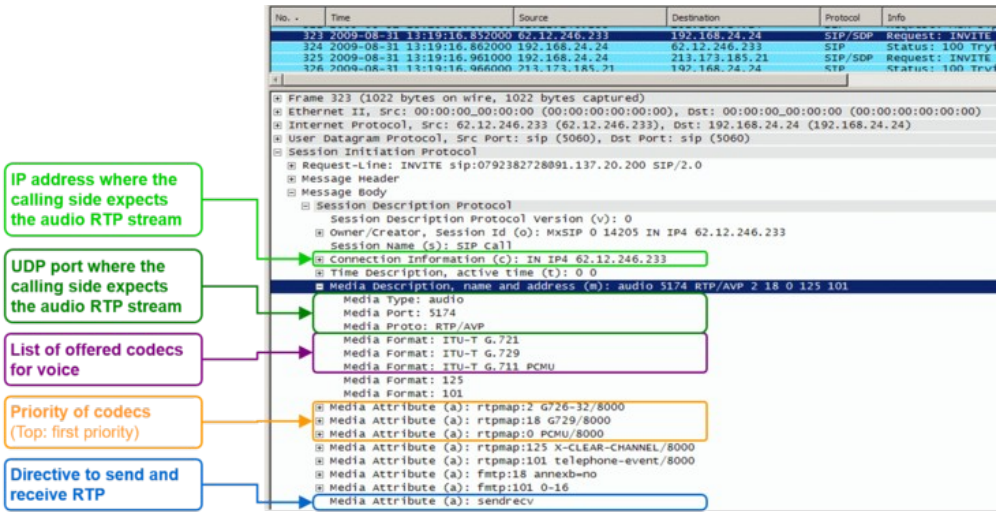
The SDP is embedded in the SIP messages during connection setup or connection renegotiation:



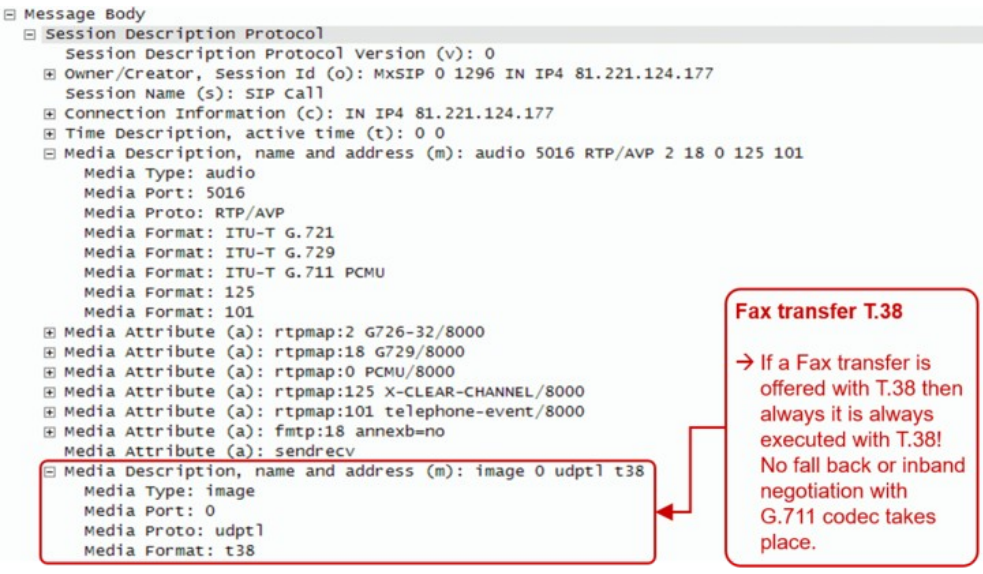
The following SDP properties and parameters are important for supporting customer problems:



Example of a SDP offer from the calling side A:



Example of a SDP offer for a Fax transfer with T.38 from the calling side A:



Interpretation of the "Media Attributes":

Index	Type	Attribute	Remark
0	PCMU	ISDN G.711µlaw	Very good quality VoIP codec
8	PCMA	ISDN G.711alaw	Very good quality VoIP codec
2	G.726-32		Good quality VoIP codec
18	G.729		Low quality VoIP codec

125	x-clear-channel	data service bit transparent	Echo canceling will be switched off and the data bit by bit transferred
101	telephone-event	DTMF, RFC 2833	DTMF will not be transferred inband but as RTP event according RFC 2833
18	annexb=0	Special information for codec with index 18	Special directive for codec G.729
101	0-16	Special information for for telephone-event with index 101	0-15 : DTMF character 0-9, *,#, A,B,C,D 0-16 : DTMF character 0-9, *,#, A,B,C,D, Flash

Basics: RTP/RTCP

The Real Time Protocol RTP is used to transfer media data, e.g. speech in VoIP based telephony.

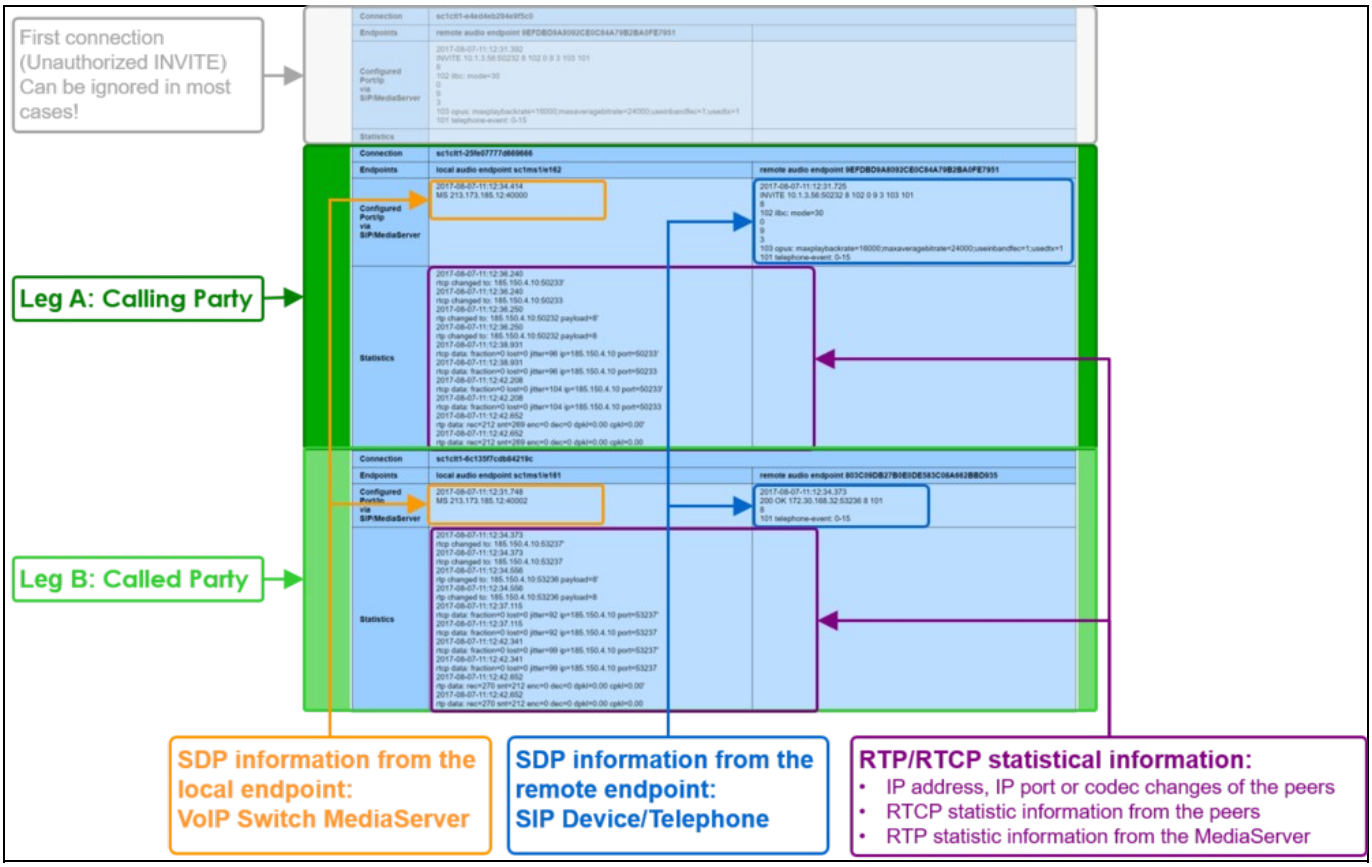
The Real Time Control Protocol RTPC transfers periodically statistical media data between the peers of a connection.

If RTP packets are lost, delayed or jitter then we speak of a Quality of Service QoS problem. For the support it is of interest to know if the number of transferred packets between the peers of a connection and if the numbers in the receive and send paths are reasonable equal, if packets were lost on call leg etc. With these statistical media information it can be possible to identify a path or transfer direction were QoS problems occure.

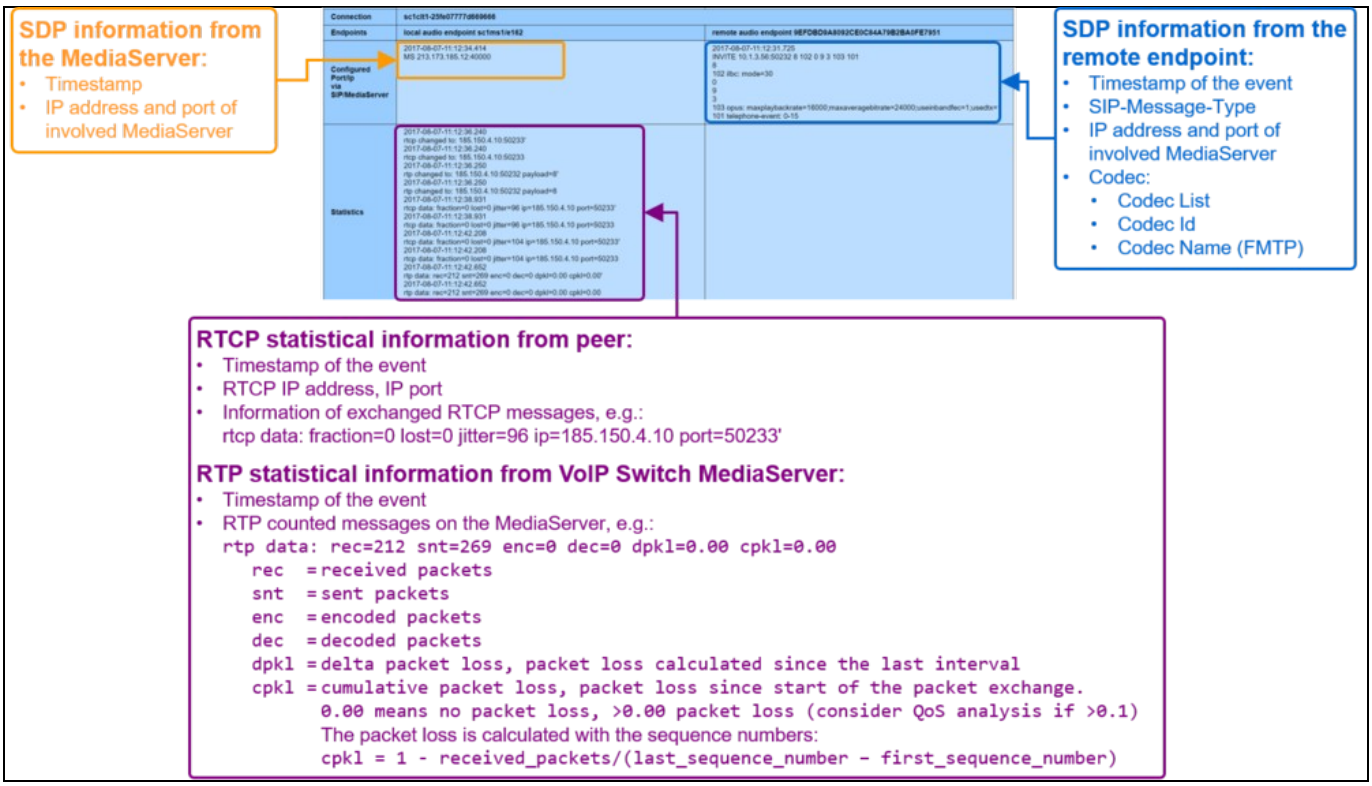
Note	The media stream must be proxied via the MediaServer of the VoIP Switch in order to compute statistical numbers of a connection.

The Aarenet VoIP Switch supports RTP/RTCP statistic data collection of a connection. How they can be obtained is described in article "Manual of the Aarenet VoIP Switch Support Tools", chapter "The ConfigCenter Call Data"

Overview of "RTP/RTCP" information collection:



Details of "RTP/RTCP" information collection:



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