



Guide of configuration and use of QoS VoIP

This guide applies from the release 4.3.0B02

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INTRODUCTION

Why QoS of the VoIP?

VoIP requires network specificities and sip-trunk to provide quality telephone conversations.

If, when setting up a network, respecting the rules of builders ensures quality of the service, the development of network infrastructure and its use require to verify if the quality of conversations is still correct.

The QoS of VoIP solution of COGIS NETWORKS can measure this by the quality indicators of QoS (Quality of Service) which are a log of telephone conversations and analysis of technical indicators .

Then the administrator has a tool to :

- measure quality.
- be alerted in case of problems.
- be able to identify the types of problems conversations.

The analysis of the QoS of VoIP is available in two packages :

- as an option of the software " Visual Taxe Pro "
- software " VTP SIP Analyzer "

This document applies to both packages.

To find out which package you're setting up, check the license certificate containing the name of the product.

Architecture and features

The software of COGIS NETWORKS is a solution running with Windows or Linux, to analyze the QoS SIP / IP terminals compatible RTCP-XR.

The operator accesses the software via a web interface.

At the end of every phone conversation, the terminal transmits a frame RTCP-XR (RFC-3611) to the software, that stores it. The part of the software that collects frames is the sip service.

According to a QoS profile, the software performs QoS monitoring and sends alerts by email if a QoS criterion is not met.

Dashboards can be triggered manually or automated with email sending, allowing the analysis of QoS according to different criteria and to a defined perimeter SIP.

An Expert Module allows analysis by terminal and called / calling number, integrated a visualization of each indicator and sub-indicator of QoS.

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FIRST STEP

To use the software you need to use a recent compliant web browser (Microsoft Internet Explorer, Mozilla Firefox or Google Chrome).

The url to enter in the browser depends on the server name or IP address, and type of installation. Ask your network administrator or your dealer.

If you connect from the server itself, you can enter the following url:

<https://localhost:8889/VisualTaxeWeb>

<http://localhost:8888/VisualTaxeWeb>

Otherwise, use the url above by replacing "localhost" with the name or IP address of the server.

You get the following screen.

Use the **user** and **password** fields to login.

Respect uppercase and lowercase.

The login of the manufacturer is **installateur**, and the password **super**.



Then click on one of the icons below:



Gives access to the configuration of the software, the look up of the newspapers and the look up of alarms of monitoring.

Gives access to the directory.

Gives access to the management of the instrument dashboards.

Gives access to the Expert Module.



At any moment, to return to this homepage, click on the image indicating the name of the product located on the left top.

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SET UP PROCEDURE

Package “Visual Taxe Pro”

If you hold the package Visual Taxe Pro, follow the following stages:

Stage		Guide to be consulted
A	Activate the license of the software.	Refer you to the paragraph “Register license” of this document.
B	Set the software (collecting tickets, importing directory).	Refer you to the guide of general configuration or the guide of simplified commissioning.
C	Follow the stages of QoS configuration.	Refer you to the paragraph “Stages to follow” of this document.

Package “VTP SIP Analyser”

If you hold package VTP SIP Analyser:

Stage		Guide to be consulted
A	Activate the license of the software.	Refer you to the paragraph “Register license” of this document.
B	Import directory LDAP or since a CSV file, or seize it manually.	Chapter “Import directory” of the guide of general configuration.
C	Follow the stages of QoS configuration.	Refer you to the paragraph “Stages to follow” of this document.

Common steps to follow

Follow the following steps:

Stage		Paragraph to be followed
1	Activate the issuance of frames RTCP-XR on the terminals.	See technical note GT156.
2	Create a SIP service and make sure the proper frame reception.	Chapter “SIP Service” of this guide.
4	Set up the QoS profiles or use those predefined.	Chapter “QoS Profiles” of this guide.
5	Set up the sending of the e-mails.	Paragraph “set up the SMTP connector” of this guide.
6	Set up the monitoring and the email alerts.	Chapter “Monitoring and e-mail alerts” of this guide.
7	Program the status reports	Chapter “dashboards” of this guide.

Next, if other elements are to configure, see the chapter “Other elements of configuration” of this guide.

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SIP SERVICE

Architecture

At the end of every telephone conversation, compatible RTCP-XR SIP and IP terminals transmit a frame that provides indicators of QoS.

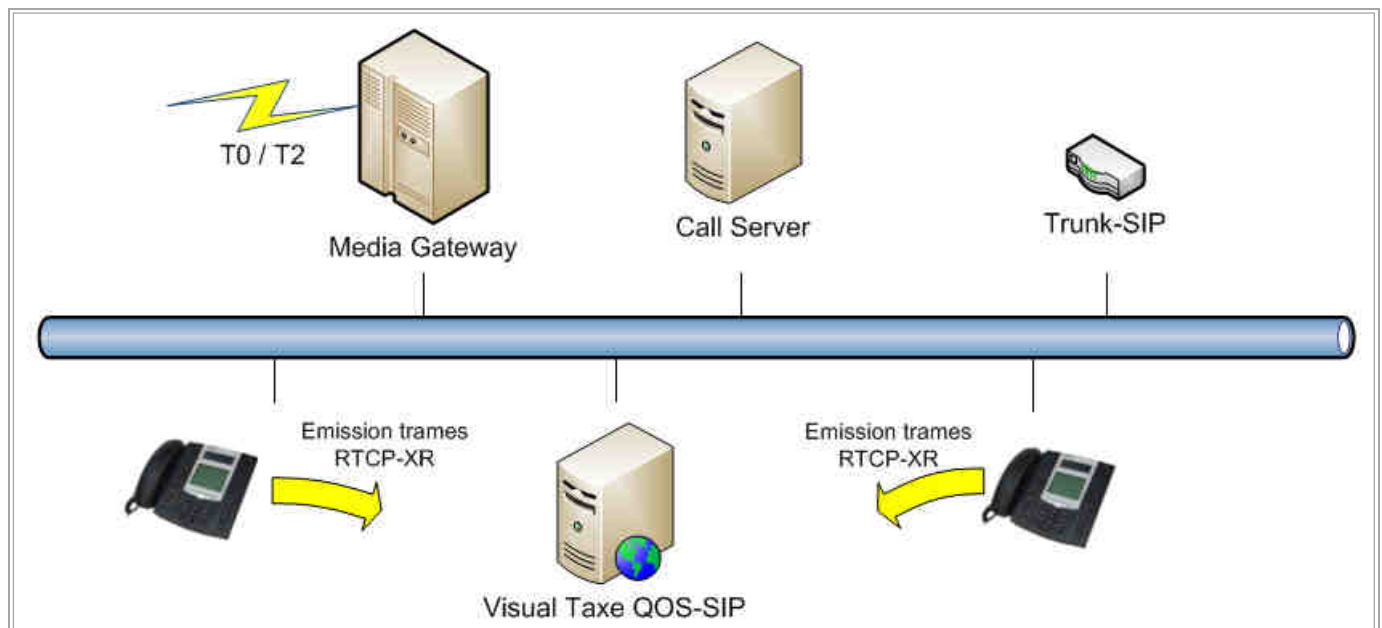
This frame is sent to the software QoS of VoIP.

Each terminal transmits its own frame, this means that:

- For a call between 1 compatible terminal and another not compatible: 1 single frame will be issued.
- For a call between 1 compatible terminal and another compatible terminal : 2 frames will be issued.

For a terminal to issue such a frame, it must:

- Be compatible RTCP-XR.
- Have been configured to send frames to the software of QoS VoIP analysis.



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What is a "SIP service"?

To collect RTCP-XR frames, the software is tuned to the latter by one or more SIP services.

One sip service is listening on all local addresses on a single server and UDP port defined in the configuration of sip service.

It is possible to have several sip services, each one listening on a different port.

Handle SIP services

Access to the management of the sip services

To access the management of sip services from the login page, enter your user name and password, then click the icon configurator.



When you have access to the configurator, left in the configuration tab, click on the SIP component then SIP Service.



On the right, you get the data grid where the sip services created appear.



Above the data grid, select the source of acquisition concerned.

Create a SIP Service

If you already have a sip service created and if you need only one, change the existing sip service instead of creating another.

To create a sip service, click the button , a new row appears in the data grid.

Write the title of the sip service to create, and then click the button in **protocol**.

A window appears, enter the UDP port on which the SIP service will be listening frames.

Then click the button



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Once you have returned to the list of SIP services, click the button



So that your action is taken into account, you must restart the SIP service process (refer to section "restart process").

Modify an existing SIP service

To modify a SIP Service:

- To change the title, click the title and enter the new one.
- For the listening port, click the button in the protocol, enter the new UDP listening port.

Then click the button



So that your action is taken into account, you must restart the service sip process (refer to section "restart process").

Delete an existing SIP service

Within the data grid, on the left of the SIP service to delete, select the checkbox and click

the button

0-Source principale			
Collecte des données - Service SIP			
<input type="checkbox"/>	Compteur	Libellé	Protocole
<input checked="" type="checkbox"/>	1	LAN Boissy	UDP: Serveur (Port:777)

So that your action is taken into account, you must restart the SIP service process (refer to section "restart process").

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QoS PROFILE

What is the purpose of QoS profiles?

For principal indicators (MOS or R-Factor, jitter, loss, signal, transit), the profiles of QoS make it possible to define the method of calculating and the acceptable values of QoS.

We determine whether using the MOS or R-Factor within the setting of the profiles.

By default, two profiles are provided, one for alerts and one for dashboards.
The operator can easily modify or create new ones.

The QoS profiles are essential because they are used:

- In the case of monitoring for sending alerts by email.

Each monitor will be established on a profile, it is possible to have multiple profiles for different monitorings.

- In the case of dashboards, in order to define the quality of conversations.

Each dashboard is based on a profile.

One profile at least is required.

Handle profiles

Access to the management of the profiles

To access to the management of QoS profiles from the login page, enter your user name and password, then click the icon configurator.



When you have access to the configurator, on the left in the **configuration** tab, click on **SIP** component then **Configuration of QoS profiles**.



On the right, you get the data grid where the created profiles appear.

Configuration of QOS profiles							
<input type="checkbox"/>	Label	MOS/R-Factor	Transit	Jitter	Loss	Signal	Maximum
<input type="checkbox"/>	Editions	<input type="text" value="MOS/R-Factor"/>	<input type="text" value="Transit"/>	<input type="text" value="Jitter"/>	<input type="text" value="Loss"/>	<input type="text" value="Signal"/>	<input type="text" value="Configuration"/>
<input type="checkbox"/>	Alertes	<input type="text" value="MOS/R-Factor"/>	<input type="text" value="Transit"/>	<input type="text" value="Jitter"/>	<input type="text" value="Loss"/>	<input type="text" value="Signal"/>	<input type="text" value="Configuration"/>

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Create a profile

In order to create a profile, click on the button  , a new line appears within the grid of data.


Indicate the title then inform each button located on the right. For that, follow the indications of the paragraph "Parameters of the profiles".

Then click on the button  .

Modify a profile

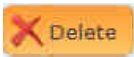
In order to modify a profile:

- To modify the title, click on the title and seize the new one,
- For other parameters, click on the button concerned and carry out the necessary modifications while following the indications of the paragraph "Parameters of the profiles".

Then click on the button  .

To remove a profile

Within the data grid, on the left of the profile to delete, select the checkbox

and click the button  .

Configuration of QoS profiles					
<input type="checkbox"/>	Label	MOS/R-Factor	Transit	Jitter	Loss
<input type="checkbox"/>	Editions	MOS/R-Factor	Transit	Jitter	Loss
<input type="checkbox"/>	Alertes	MOS/R-Factor	Transit	Jitter	Loss

Settings of profiles

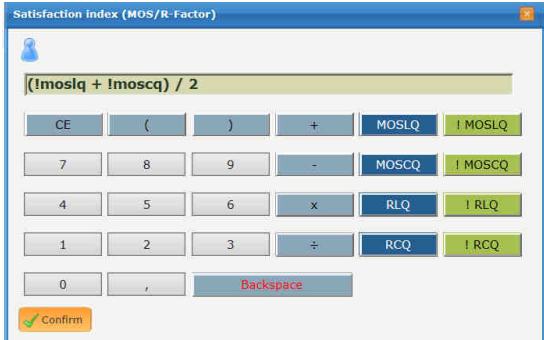
Calculation of the main indicators

If you click one of the buttons (MOS, jitter, etc ...), you get this calculator.

The purpose is to define the method of calculation of indicators.

It is strongly recommended not to change these parameters without a good understanding of the software and QoS indicators of VoIP.

Within one single frame RTCP-XR, different sub-indicators are provided (see " QoS indicators "), it is necessary to determine from which sub-indicators will be calculated the main indicators.



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For each indicator, you must specify the formula, according to the rules of priority of multiplication / division before addition / subtraction, with management brackets.

The sub-indicators available are shown on the right, in our example are the abbreviations MOSQL, MOSLQ, MOSCQ, MOSCQ, etc ...
For abbreviations, see the section "QoS indicators."

Each sub-indicator is present 2 times, once with a previous question mark and once without.
The presence of the question mark means that only meaningful values will be taken into account, without the question mark all values will be taken into account.



thresholds

Within the setting of thresholds, are defined for each main indicator value ranges that identify thresholds:

- Excellent
- Good
- Reasonable
- Low
- Very low

To change values, click the label of one of the ranges, and slide the label, while keeping the mouse button pressed until the desired value, which one is automatically updated as and when displacement.



The buttons   located on both sides, allow to change the maximum and minimum values.

It is also within this screen, on the top left, that we will chose if the analysis is performed on the MOS or the R-Factor.

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SURVEILLANCE & EMAIL ALERTS

What is the surveillance for ?

The software is able to generate alerts in case of exceeding a threshold indicator.
For this, a surveillance is implemented.

When monitoring, if one or more indicators have a value below the threshold defines, an alert can be sent by email.

One surveillance relates to one type of conversation and one alert threshold.

Handle surveillance

Access to the management of the surveillances

To access to the management of surveillances from the login page, enter your user name and password, then click the icon configurator.



When you have access to the configurator, on the left in the **configuration** tab, click on SIP and then on **SIP surveillance**.



On the right, you get the data grid where surveillance created appear.

SIP Surveillance					
<input type="checkbox"/>	Activation	Label	QOS profile	Analyze SIP	Period of analysis
<input type="checkbox"/>	true	cogis-lan	Alertes	Click here	Click here

Create a surveillance

In order to create a surveillance, click on the button , a new line appears within the grid of data.

Inform each one of the fields and button located on the right. For that, follow the indications of the paragraph "Parameters of the surveillances".

Then click on the button



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Modify a surveillance

In order to modify a surveillance :

- To change the title, click the title and enter the new one,
- For other settings, click the button concerned and make the required changes by following the indications of the paragraph "Parameters of surveillances".

Then click the button



Remove a surveillance

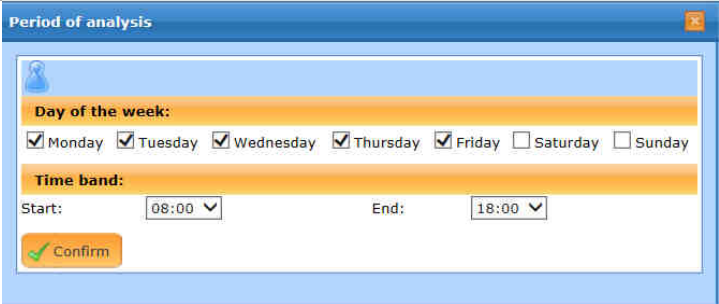
Within the data grid to the left of the monitor to delete, select the checkbox and click the button



SIP Surveillance				
<input type="checkbox"/>	Activation	Label	QoS profile	Analyze SIP
<input type="checkbox"/>	true	cogis-lan	Alertes	Click here

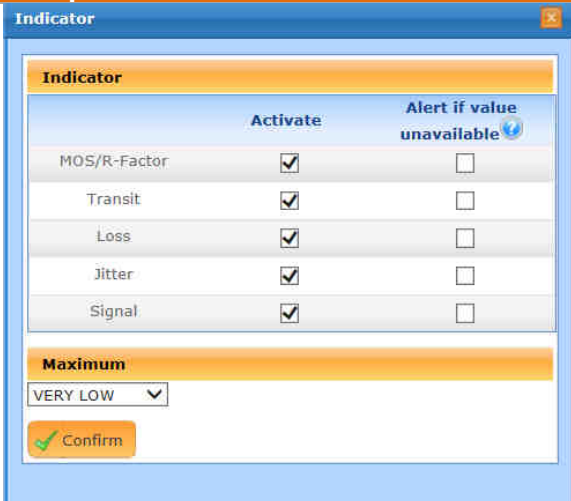
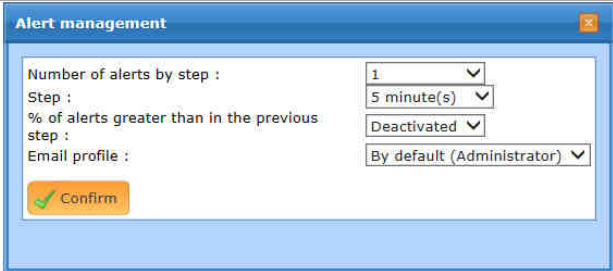
Settings of surveillance

The SIP surveillances are defined according to the criteria of the table below.

Element	Settings to provide	
Activation	Allows to indicate if a SIP surveillance is active or not. If a surveillance is inactive, no analysis nor sending of e-mail will be carried out according to the criteria of the surveillance.	
Title	Title of the surveillance.	
QoS Profile	Defines the QoS profile to use for this surveillance (see the paragraph "QoS Profiles").	
SIP Analysis	Defines the perimeter of surveillance (see the paragraph "perimeter of sip analysis").	
Period of analysis	Allows to define during which days of the week and which time slots the surveillance is active.	

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Element	Settings to provide	
Threshold indicator	<p>This screen allows you to define which QoS indicators to take into account in the surveillance and what is the threshold at which an alert is set.</p> <p>The column Alert if value unavailable force the sending of an alert if the value of the indicator in question is unavailable in the frame RTCP-XP.</p>	
Management of alerts	<p>Alert management can limit the number of alerts issued for the same surveillance, defining:</p> <ul style="list-style-type: none"> - frequency of sending alerts (in minutes) - the maximum number of alerts per frequency - the percentage of overshoot required since the last alert - the mailing list to which the email is to send 	

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Sending email alerts

Alerts are sent by mail via an SMTP connector to the mailing lists called "emails profiles".

To send alerts by email, you must first:

- Configure the SMTP connector for this, see the section "Set up the SMTP connector."
- Define profiles emails, see section "Emails profiles"

Finally, for an alert to be sent by email, see section "Parameters of surveillances."

Consult alerts on the screen

To access the alerts issued, from the login page, enter your user name and password, then click the icon configurator.



When you have access to the configurator, on the left in the **tools** tab, click **Logbooks** and then **SIP alert reports**.



- Collector
- Remote collector
- Planner
- Room management
- WEB
- SIP service
- SIP alert report
- Login log

On the right, you get the data grid where alerts appear.

Above, select the surveillance for which you want to see the alerts:



SIP alert report							
<input type="checkbox"/>	Date	Extension	Satisfaction index (MOS)	Transit time (milliseconds)	Data loss (%)	Phase jitter (milliseconds)	Quality of audio signal (dbm)
<input type="checkbox"/>	23/09/2013 20:34:19	124	4,40 (Good)	71,00 (Excellent)	0,00 (Excellent)	0,00 (Excellent)	-12,00 (Good)
<input type="checkbox"/>	23/09/2013 20:34:18	100	4,10 (Good)	132,00 (Excellent)	0,00 (Excellent)	18,00 (Excellent)	-4,00 (Good)
<input type="checkbox"/>	23/09/2013 20:34:17	144	4,20 (Good)	94,00 (Excellent)	0,00 (Excellent)	0,00 (Excellent)	-24,00 (Excellent)
<input type="checkbox"/>	23/09/2013 20:34:17	119	4,30 (Good)	88,00 (Excellent)	0,00 (Excellent)	0,00 (Excellent)	-21,00 (Excellent)
<input type="checkbox"/>	23/09/2013 20:34:17	119	4,30 (Good)	96,00 (Excellent)	0,00 (Excellent)	0,00 (Excellent)	-15,00 (Excellent)
<input type="checkbox"/>	23/09/2013 20:34:17	119	4,30 (Good)	67,00 (Excellent)	0,00 (Excellent)	0,00 (Excellent)	-20,00 (Excellent)
<input type="checkbox"/>	23/09/2013 20:34:17	119	4,10 (Good)	59,00 (Excellent)	0,00 (Excellent)	0,00 (Excellent)	-18,00 (Excellent)

If you forgot the parameters which compose a surveillance, above the grid of data, click on



, and below you will obtain a reminder of the parameters.

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STATUS REPORTS

General information

The software has 5 status reports allowing to analyze the quality of conversations.

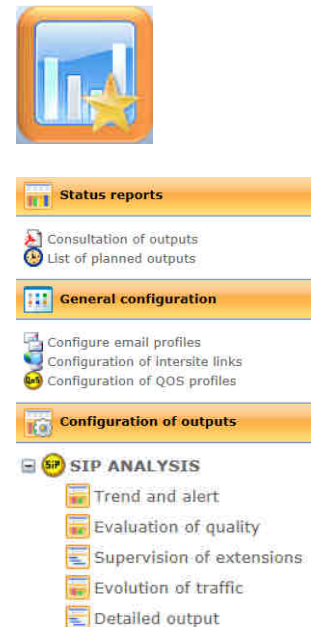
Status reports can be performed manually or automatically and received by email.

Handle status reports

Access to the management of status reports

To access to the management of status reports from the login page, enter your user name and password, then click the dashboards.

Then you have access to the management of status reports.



See published status reports

On the left, click **consultation of outputs**.

On the right, you get the data grid where outputs have been published.

To see the status report, just click the icon in the download column that is published on the status report you want to see.


Access configuration on status reports

On the left, in **configuration of outputs**, within the component **SIP ANALYSIS**, click the template of status report that you want to program.

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Create a status report

To create a status report, after selecting the template desired, click the button  , a new row appears in the data grid.

Complete each of the fields and buttons on the right. For that, see the paragraph "Settings status reports."

Then click the button



Edit a status report

To edit a status report, after selecting the template:

- To change the title, click the title and enter the new one,
- For other settings, click the question and make the required changes by following the indications under "Configuration of status reports" button.

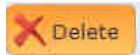
Then click the button



Remove a dashboard

Within the data grid, after selecting the template, on the left of the status report to delete, select the checkbox and click the

button



Trend and alert				
<input type="checkbox"/>	Label	Frequency	Period of analysis	Analyze SIP
<input type="checkbox"/>	Table 8	None	Period of analysis	Click here
<input type="checkbox"/>	Table 9	None	Period of analysis	Click here

Immediately execute a status report

After setting up the status report, set the analysis period and then click **Execute immediately** on the line of the status report.

Wait 1 minute, then go to "Consultation of outputs" in order to see the status report.

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Parameters of status reports

According to the template, the parameters available can differ.

Element	Settings to provide		
Title	Title of the report that also appears on the document.		
Frequency	Frequency of the planning of the report.		
	Frequency	Coming out date	Analyzed Period
	None	According to manual release	According to manual configuration
	Once a day	Everyday	Day before the release date
	Weekly	Each monday	Week before the release date
	Monthly	The 1st of each month	Month before the release date
	Bimestrial	The 1st of each odd month	2 months before the release date
Period of analysis	The period of analysis is made up of several parts:		
	Parameter	Description	
	Date/time	Allows you to specify the date / time of beginning and end of frames to consider. This parameter is used only when the report is triggered manually. If the report automatically emerges, the period is calculated from the frequency.	
	Day of the week	Days of the week to take into account.	
	Time band	Time range to consider.	
	Breaking time band	Breaking time to insert in the schedule.	
Analyze SIP	Allows to define which conversations must be analyzed. See “Scope of analysis” of this guide.		

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Element	Settings to provide	
Parameters	According to the template, available parameters are different:	
	Parameter	Description
	E-mail	Profile email to which the email will be sent. If you specify "none", the report will not be sent by email then you can only view or print it.
	Public	If checked, all user accounts have access to the report (but not to the configuration of the report).
	QoS Profile	QoS profile to use.
	Distribution	Indicates on what information data will be accumulated.
Execute immediately	Clicking this button triggers the immediate execution of the status report according to the specified settings. The analysis period will be the one specified in Analysis period , regardless of frequency.	
Output mode	According to the template, you can choose between PDF or XLS.	

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Description of status reports

Model "Trend and alert"

This dashboard provides as a PDF file, a summary of the quality of telephone conversations.

Are supplied :

- Table with the score and the qualitative results of key indicators: MOS, transit, jitter, loss and quality of signal.
- Pie chart providing, according to each type of flows (incoming + outgoing / internal combined), the MOS as a score and qualitative result.
- Chart (histogram) providing MOS per daypart, as a score and qualitative result.

Model "Evolution of traffic"

This dashboard provides, as a PDF file, charts about the evolution of each one of the key indicators (MOS, transit, loss, jitter) per frequency of 5 minutes.

Model "Supervision of extensions"

This dashboard provides, as a PDF file, a top 5 of the terminals per indicator (MOS, transit, loss, jitter, signal) with the worst score.

The terminals are indicated by their number or by their IP address.

Model "Evaluation of quality"

This dashboard provides, as a PDF file, an assessment of various indicators (MOS, transit, jitter, loss) during a period in the form of totals by time slot (60, 30 or 15 minutes), day of week, day of month, week, month.

Model "Detailed output"

This dashboard provides, as an XLS file, the list of the frames RTCP-XR for the selected place.

Data provided are the ones indicated in paragraph "Counters RTCP-XR."

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Examples

“Trend and alert”



Table8
Trend and alert

VISUAL TAXE PRO

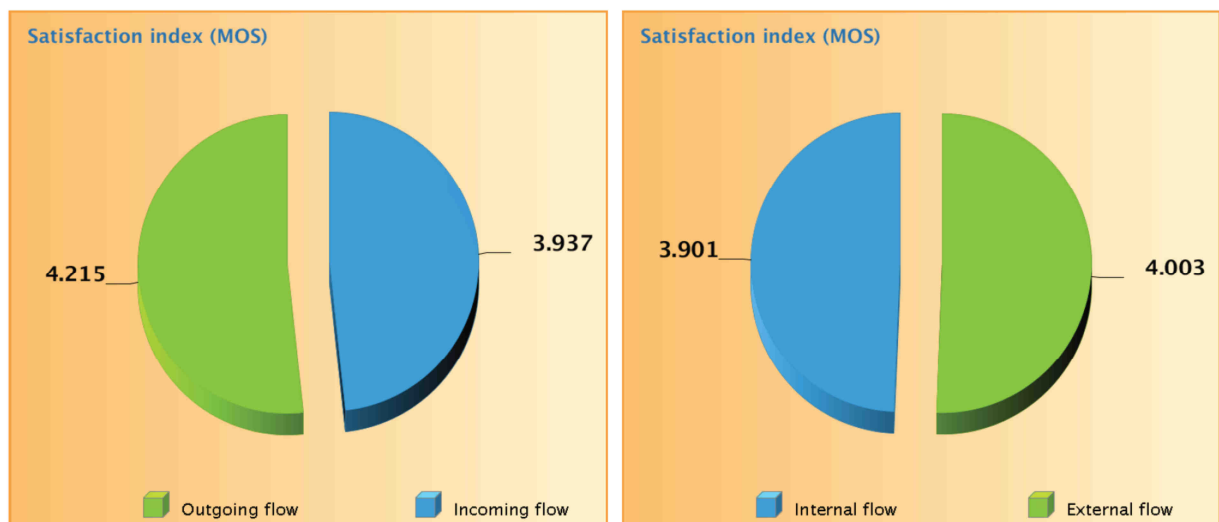
Du 6/28/13 8:12:39 AM au 7/3/13 6:55:18 PM entre 12:00:00 AM et 11:59:59 PM

Summary of the service quality of SIP communications

COGIS-QOS

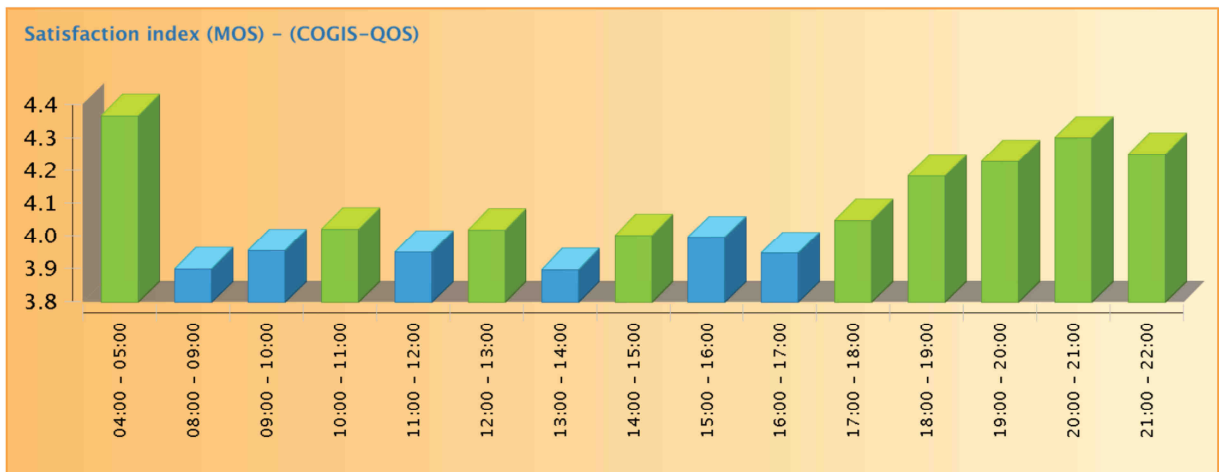
Indicator	Score	Result
Satisfaction index (MOS)	3,99	REASONABLE
Transit time (milliseconds)	103 ms	EXCELLENT
Phase jitter (milliseconds)	19 ms	EXCELLENT
Data loss (%)	0.28%	EXCELLENT
Quality of audio signal (dBm)	-15.13 dBm	GOOD

NB: The satisfaction index measures the quality of a telephone conversation



Outgoing flow = outgoing calls or internal calls made, Incoming flow = Incoming calls or internal calls received
Internal flow = internal calls made or received, External flow = outgoing calls or incoming calls

Key	EXCELLENT	GOOD	REASONABLE	LOW	VERY LOW
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“Evolution of traffic”

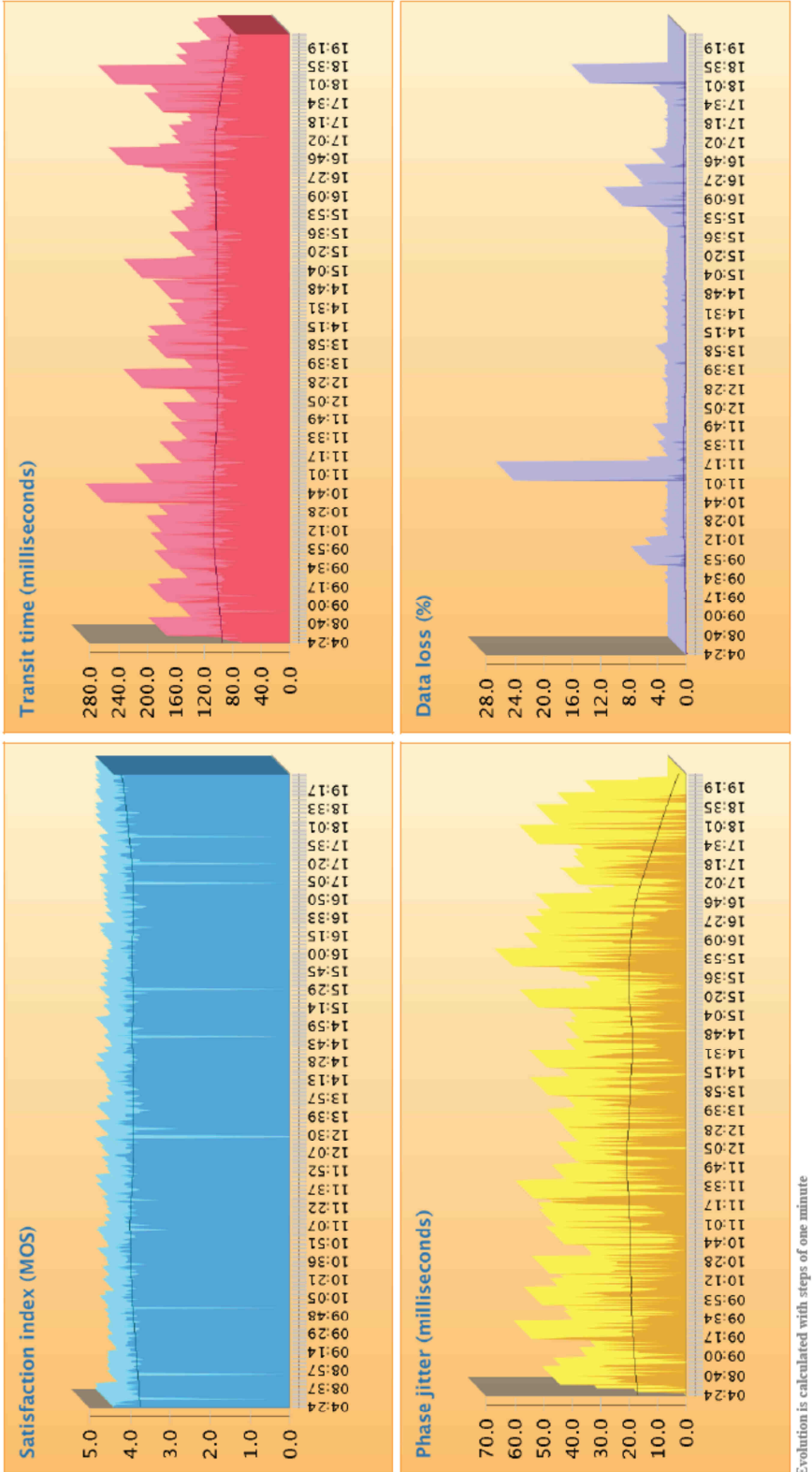
VISUAL TAXE PRO

Table11
Evolution of traffic

Du 6/28/13 8:12:39 AM au 7/3/13 6:55:18 PM entre 12:00:00 AM et 11:59:59 PM



Evolution of SIP traffic during the period
COGIS-QOS



Guide of configuration/use of QoS VoIP

This guide applies from the release 4.3.0B02

“Supervision of extensions”



VISUAL TAXE PRO

Table10
Supervision of extensions

Du 6/28/13 8:12:39 AM au 7/3/13 6:55:18 PM entre 12:00:00 AM et 11:59:59 PM

cogis-sip

Classification of extensions with the lowest satisfaction index (MOS)

Extension	Surname	Forename	Score	Result
103	Poste 103		3,75	REASONABLE
147	Poste 147		3,84	REASONABLE
138	Poste 138		3,85	REASONABLE
221	Poste 221		3,85	REASONABLE
106	Poste 106		3,86	REASONABLE

Classification of extensions with the lowest transit time (milliseconds)

Extension	Surname	Forename	Score	Result
101	Poste 101		66 ms	EXCELLENT
ANONYMOUS			68 ms	EXCELLENT
127	Poste 127		69 ms	EXCELLENT
154	Poste 154		73 ms	EXCELLENT
148	Poste 148		75 ms	EXCELLENT

Classification of extensions with the lowest phase jitter (milliseconds)

Extension	Surname	Forename	Score	Result
145	Poste 145		0 ms	EXCELLENT
112	Poste 112		0 ms	EXCELLENT
129	Poste 129		0 ms	EXCELLENT
121	Poste 121		0 ms	EXCELLENT
127	Poste 127		0 ms	EXCELLENT

Classification of extensions with the highest data loss (%)

Extension	Surname	Forename	Score	Result
145	Poste 145		0%	EXCELLENT
121	Poste 121		0%	EXCELLENT
129	Poste 129		0%	EXCELLENT
103	Poste 103		0%	EXCELLENT
155	Poste 155		0%	EXCELLENT

Classification of extensions with the lowest audio signal quality (dBm)

Extension	Surname	Forename	Score	Result
127	Poste 127		-25.71 dBm	EXCELLENT
155	Poste 155		-23.33 dBm	EXCELLENT
145	Poste 145		-22.57 dBm	EXCELLENT
ANONYMOUS			-22.16 dBm	EXCELLENT
132	Poste 132		-21.85 dBm	EXCELLENT

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This guide applies from the release 4.3.0B02

“Evaluation of quality (page 1)”

VISUAL TAXB PRO



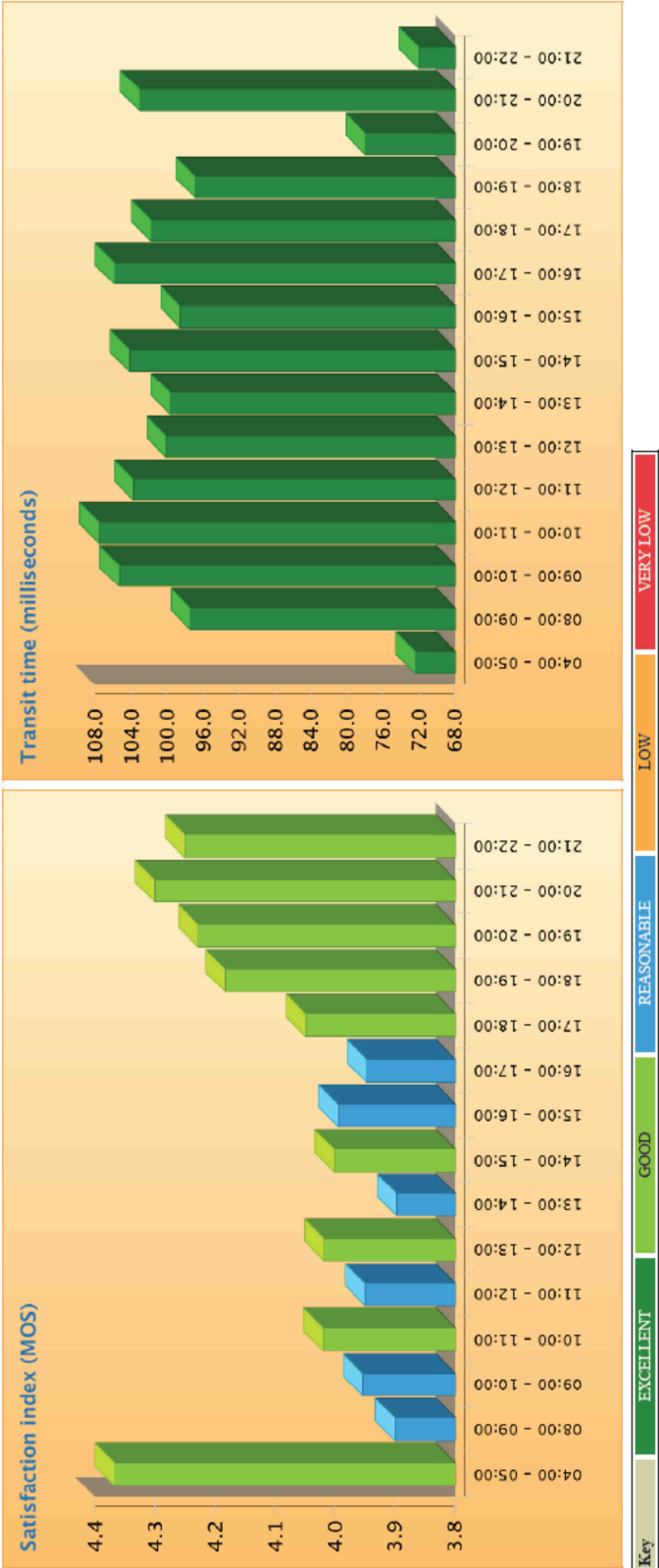
Table9
Evaluation of quality

Du 6/28/13 8:12:39 AM au 7/3/13 6:55:18 PM entre 12:00:00 AM et 11:59:59 PM

Hourly distribution of service quality of SIP communications
cogis-sip

Indicator	Score	Result
Satisfaction index (MOS)	3.99	REASONABLE
Transit time (milliseconds)	103 ms	EXCELLENT

NB: The satisfaction index measures the quality of a telephone conversation



This guide applies from the release 4.3.0B02

“Evaluation of quality (page 2)”

VISUAL TAXE PRO

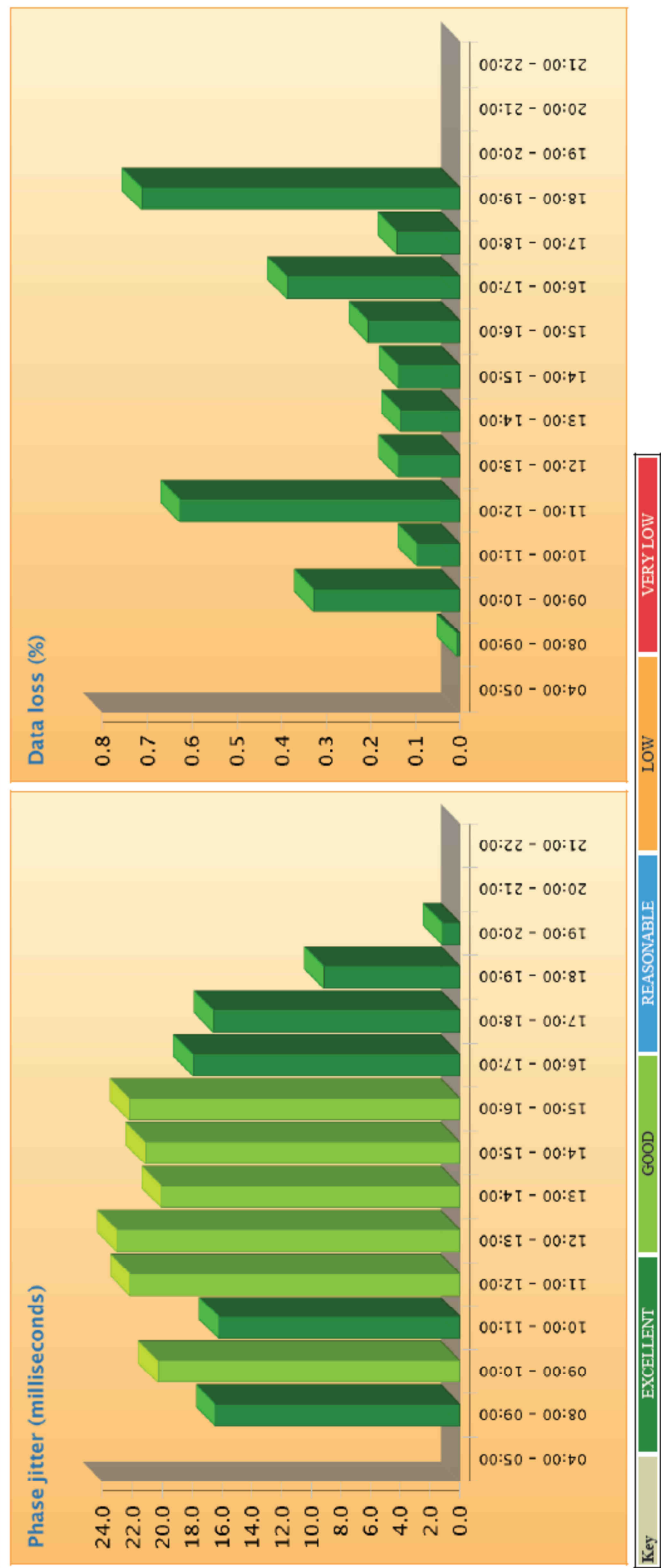
Table9
Evaluation of quality

Du 6/28/13 8:12:39 AM au 7/3/13 6:55:18 PM entre 12:00:00 AM et 11:59:59 PM

Hourly distribution of service quality of SIP communications

cogis-sip

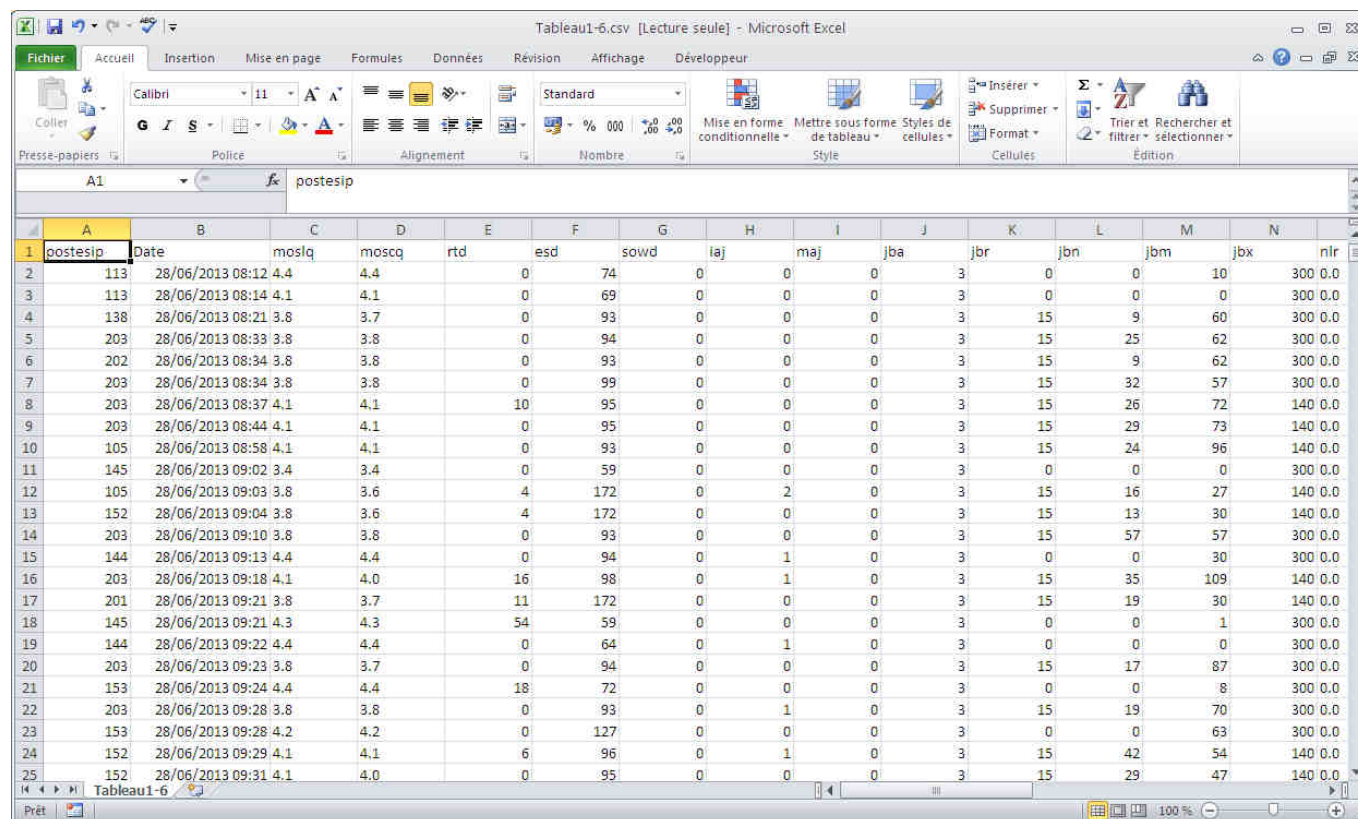
Indicator	Score	Result
Phase jitter (milliseconds)	19 ms	EXCELLENT
Data loss (%)	0.28%	EXCELLENT
Quality of audio signal (dBm)	-15.18 dBm	GOOD



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“Detailed output”



	A	B	C	D	E	F	G	H	I	J	K	L	M	N	
1	postesip	Date	moslq	moscq	rtd	esd	sowd	iaj	maj	jba	jbr	jbn	jbm	jbx	nlr
2	113	28/06/2013 08:12	4.4	4.4	0	74	0	0	0	0	3	0	0	10	300 0.0
3	113	28/06/2013 08:14	4.1	4.1	0	69	0	0	0	0	3	0	0	0	300 0.0
4	138	28/06/2013 08:21	3.8	3.7	0	93	0	0	0	0	3	15	9	60	300 0.0
5	203	28/06/2013 08:33	3.8	3.8	0	94	0	0	0	0	3	15	25	62	300 0.0
6	202	28/06/2013 08:34	3.8	3.8	0	93	0	0	0	0	3	15	9	62	300 0.0
7	203	28/06/2013 08:34	3.8	3.8	0	99	0	0	0	0	3	15	32	57	300 0.0
8	203	28/06/2013 08:37	4.1	4.1	10	95	0	0	0	0	3	15	26	72	140 0.0
9	203	28/06/2013 08:44	4.1	4.1	0	95	0	0	0	0	3	15	29	73	140 0.0
10	105	28/06/2013 08:58	4.1	4.1	0	93	0	0	0	0	3	15	24	96	140 0.0
11	145	28/06/2013 09:02	3.4	3.4	0	59	0	0	0	0	3	0	0	0	300 0.0
12	105	28/06/2013 09:03	3.8	3.6	4	172	0	2	0	0	3	15	16	27	140 0.0
13	152	28/06/2013 09:04	3.8	3.6	4	172	0	0	0	0	3	15	13	30	140 0.0
14	203	28/06/2013 09:10	3.8	3.8	0	93	0	0	0	0	3	15	57	57	300 0.0
15	144	28/06/2013 09:13	4.4	4.4	0	94	0	1	0	0	3	0	0	30	300 0.0
16	203	28/06/2013 09:18	4.1	4.0	16	98	0	1	0	0	3	15	35	109	140 0.0
17	201	28/06/2013 09:21	3.8	3.7	11	172	0	0	0	0	3	15	19	30	140 0.0
18	145	28/06/2013 09:21	4.3	4.3	54	59	0	0	0	0	3	0	0	1	300 0.0
19	144	28/06/2013 09:22	4.4	4.4	0	64	0	1	0	0	3	0	0	0	300 0.0
20	203	28/06/2013 09:23	3.8	3.7	0	94	0	0	0	0	3	15	17	87	300 0.0
21	153	28/06/2013 09:24	4.4	4.4	18	72	0	0	0	0	3	0	0	8	300 0.0
22	203	28/06/2013 09:28	3.8	3.8	0	93	0	1	0	0	3	15	19	70	300 0.0
23	153	28/06/2013 09:28	4.2	4.2	0	127	0	0	0	0	3	0	0	63	300 0.0
24	152	28/06/2013 09:29	4.1	4.1	6	96	0	1	0	0	3	15	42	54	140 0.0
25	152	28/06/2013 09:31	4.1	4.0	0	95	0	0	0	0	3	15	29	47	140 0.0

This guide applies from the release 4.3.0B02

EXPERT MODULE

General information

The Expert module allows a more thorough analysis of QoS of the VoIP, in particular by analyzing the sub-indicators of QoS.

The user can analyse QoS of each terminal, in globally, according to the type of communication, according to the correspondent with whom the communication was carried out.

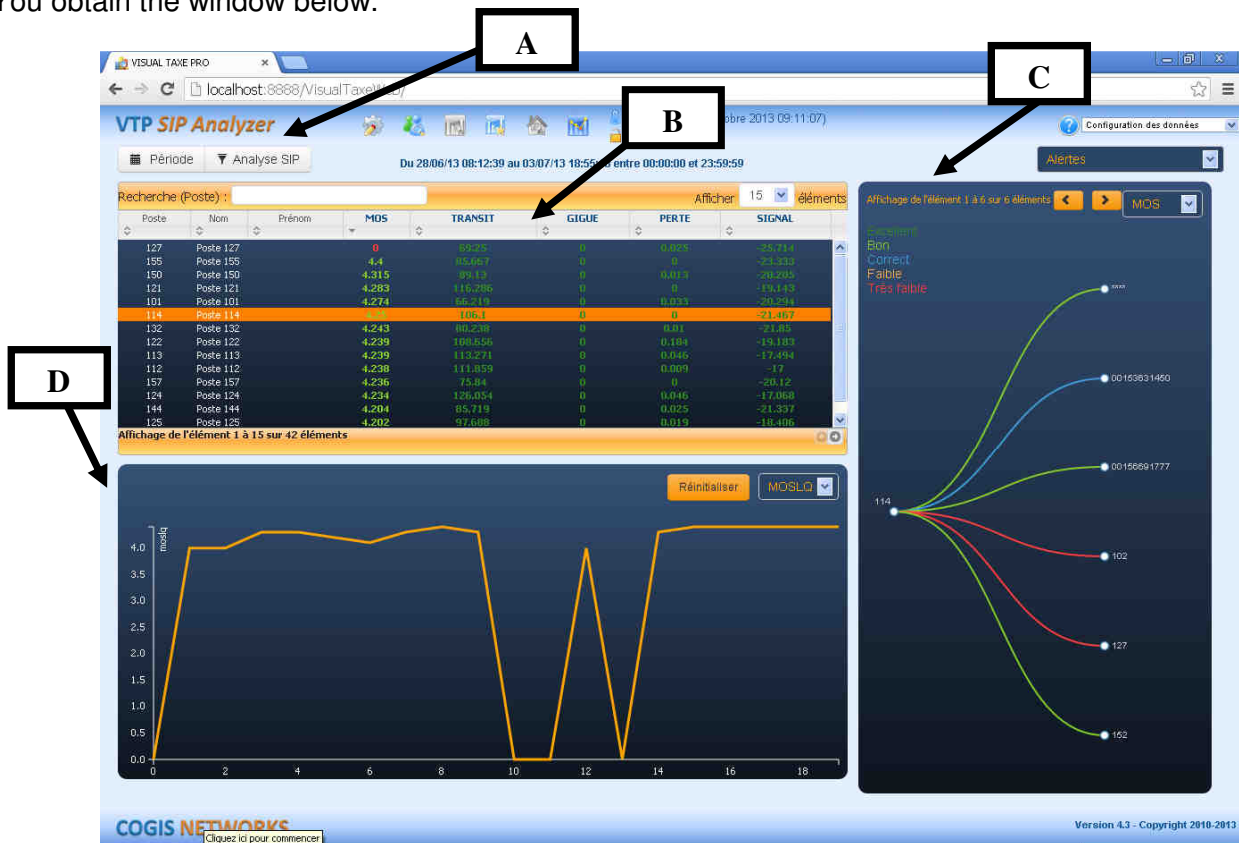
Access to the Expert Module

To access to the Expert module, since the page of identification, inform your **username** and your **password**, then click on the icon of the module:



Presentation of the interface

You obtain the window below.



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This window includes 4 areas:

Area (of the preceding page)	Use
A	This area makes it possible to select the period of analysis as well as the perimeter to be analysed.
B	Within this area, a summary of the QoS by terminal appears.
C	While clicking on a line of the table of the area B, a dendrogram is provided indicating for the selected terminal, the quality of the indicators of QoS by correspondent with whom the conversations took place.
D	While clicking on a line of the table of the area B, you obtain the evolution of a sub-indicator of QoS according to the conversations.

Description

Period of analysis

You must define the period of analysis.

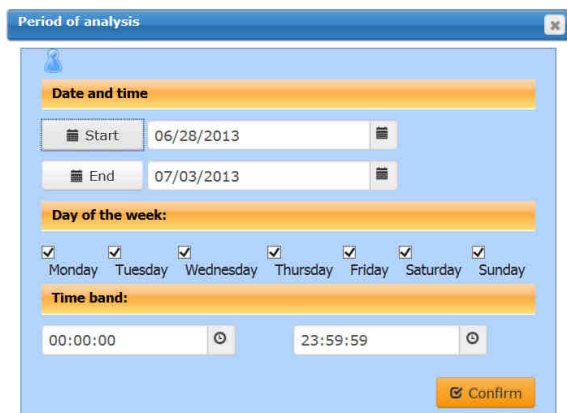
For that, it is necessary to click on the button , thus you obtain the screen below.

In “date and time”, define the date of beginning and the date of end (if you click on Start, the software will indicate the oldest date stored in the source, idem for the date of end).

In “day of the week”, you can select the days of the week to be analysed.

Lastly, in “time band” define the time band to be analysed.

Once the configuration done, click on the button



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SIP Analysis

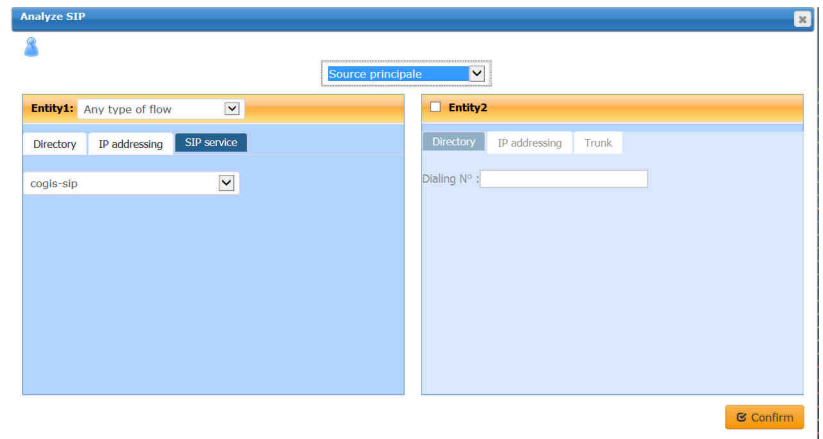
You must define the perimeter of analysis, that-is-to-say which conversations are to be analysed.

For that, in top on the left, click on



You obtain this window:

You must define the perimeter.
For that, follow the indications of the chapter "Perimeter of SIP analysis" of this guide.

A screenshot of the "Analyze SIP" configuration window. It has a "Source principale" dropdown at the top. Below are two main sections: "Entity1" and "Entity2". "Entity1" has tabs for "Directory", "IP addressing", and "SIP service", with "SIP service" selected. It contains a "cogis-sip" dropdown. "Entity2" has tabs for "Directory", "IP addressing", and "Trunk", with "Trunk" selected. It contains a "Dialing N°" input field. At the bottom right is a "Confirm" button.

Then click



Table of analysis by terminal

After having defines the period and the perimeter of analysis, the table of analysis by terminal fills itself.

A screenshot of the "Table of analysis by terminal" interface. At the top is a search bar labeled "Search (Extension) :". Below it is a table with columns: "Extension", "Surname", "Forename", "TRANSIT", "TE", and "SIGNAL". The table contains 15 rows of data. A "Show 15 entries" dropdown is at the top right. Two boxes, labeled "A" and "B", are overlaid on the table. Box "A" is over the "TE" column header, and Box "B" is over the "Forename" column header. The table shows data for various extensions, including 127, 155, 150, 121, 101, 114, 132, 122, 113, 112, 157, 124, 144, and 125. The "TRANSIT" column shows values like 0, 4.4, 4.315, 4.274, 4.25, 4.243, 4.239, 4.238, 4.236, 4.234, 4.204, and 4.202. The "TE" column shows values like 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0. The "SIGNAL" column shows values like -15.213, -13.223, -10.205, -19.143, -19.163, -17.494, -17, -20.12, -17.068, -21.337, -18.406.

Each line represents a terminal.

By defaults only 15 terminals appear, to select more terminals (limited to 300), modify the value in top on the left table (A).

The contents of the table can be sorted on a column by one 1st click on the heading of the column which causes ascending sort ; a second click causes a descending sort.

You can filter on a number of terminal.

For that, in top in the area "Search (extension)", seize whole or part of the number, then click on <Entrer>. To cancel a search, remove the seized value, then click on <Entrer>.

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Displayed data are:

Column	Meaning
Extension	Number of the terminal.
Surname and forename	Name and first name assigned in the directory to the terminal (if the directory is indicated within the software).
MOS*	Average of the MOS of all the conversations of the terminal.
DELAY*	Average of the delay of all conversations of the terminal.
JITTER*	Average of the jitter of all conversations of the terminal.
LOSS*	Average of the loss of data of all the conversations of the terminal.
SIGNAL*	Average of the signal of all conversations of the terminal.

* to obtain information on the indicators, see chapter “Gloassary of QoS” of this guide.

The colours of the indicators are related to the profile of QoS defined on the top right, that can be modified.

Editions



Clicking on a line of the table will generate 2 graphs:

- On the right: the dendrogram
- Below: evolution of the sub-indicators

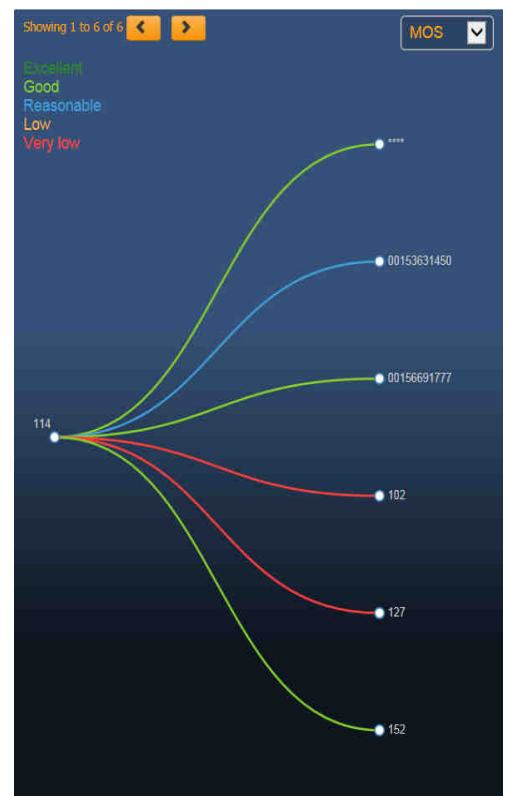
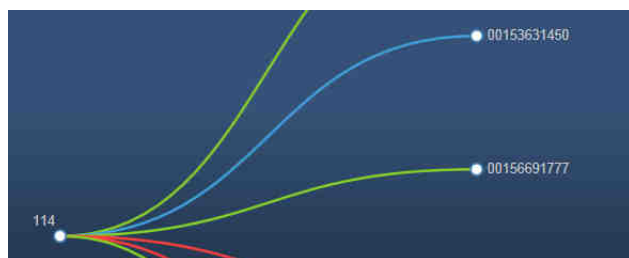
Dendrogram

The dendrogram provides per pair of phone call, that is to say selected terminal and correspondent of the phone calls, the quality of one of the indicators.

The colour indicated is related to the selected QoS profile, and represents the quality of the indicator selected in top on the right of graph.

For example, if you selected item 114 within the table, then you will be able to observe the display below in the dendrogram.

The line between “114” and “00153631450”, which is blue, provides the quality of the indicator selected for the communications between both numbers.



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If you position the slider of mouse (without clicking) on the line connecting the 2 numbers, in top on the left of the dendrogram will be indicated the number of the correspondent, the number of communications, and if the correspondent is an internal, his name.

Excellent
Good
Reasonable
Low
Very low
Dialing N° : 00153631450
Comm. : 1
Label : 00153631450

If you click on the line connecting 2 numbers, a table appears, providing for each communication the totality of the sub-indicators and complementary elements.

Communications between the extension 114 and 152



Show 15 entries

Extension	Dialing N°	Date and time	Duration	MOSLQ	MOSCQ	RLQ	RCQ	RTD	ESD	JBR	JBN	JBM	JBX	NLR	JDR	BLD	GLD	SL	NL	BD	SOWD	IAJ	MAJ	JBA	GD	GMIN	RERL	ES	Type of flow	Label	Local IP	Remote IP	PD	
114	152	6/28/13 11:48 AM	165	4	4	null	null	0	68	0	0	0	300	0	0	99,6	0	-11-68	12	935	0	0	0	3	65	535	16	75	0	1	N.CARMEL	192.168.200.122	192.168.200.222	G.711 aLaw
114	152	7/2/13 3:22 PM	175	4,4	4,4	null	null	0	113	0	0	49	300	0	0	0	0	-15-71	0	0	0	1	0	3	0	16	75	0	1	N.CARMEL	192.168.200.122	192.168.200.222	G.711 aLaw	

Showing 1 to 2 of 2

Data provided are:

Data	Meaning
Extension	Number of the selected terminal
Dialing Nr	Number of the correspondent
Date/time	Date and time of the conversation
Duration	Duration of the conversation
MOSLQ, MOSCQ, RLQ, etc...	Sub-Indicators (see chapter "Glossary of QoS" of this guide)
Type of flow	
Label	Name of the terminal such as defined within the frame of QoS
Local IP	IP address of the terminal
Remote IP	IP address of the correspondent
PD	Codec used

Icon XLS located in top on the left of the table makes it possible to carry out an export of the data of the table towards an Excel file.

Guide of configuration/use of QoS VoIP

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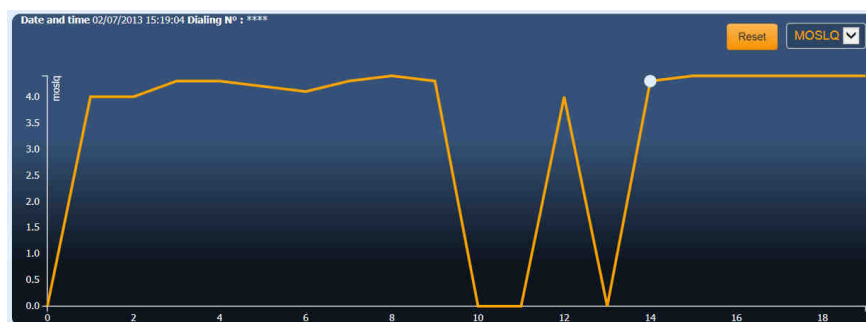
Graph of evolution of a sub-indicator

This graph provides for a sub-indicator, the evolution of the value of this indicator throughout the different conversations of the selected extension.

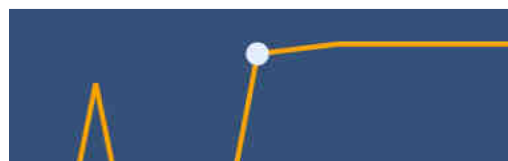
On the top right, you can select the sub-indicator to be analysed.

The Y-axis represents the value of the sub-indicator, the X-axis represents the communications.

This example provides the evolution of the MOSLQ throughout phone calls.



If you move the mouse (without clicking) on the line of evolution, you will see a white sticker as shown here.
This tablet represents the value of sub-indicator for a call.



If you click on the tablet, a table appears and provides all sub-indicators and additional elements of communication (to know the contents of the table, see paragraph about the dendrogram).

Comparateur de tickets SIP

Extension	Dialing N°	Date and time	Duration	MOSLQ	MOS	QCQ	RLQ	RCQ	RTD	ESD	JBR	JBN	JBM	JBX	NLR	JDR	BLD	GLD	SL	NL	BD	SOWD	IAJ	MAJ	JBA	GD	GMIN	RER	LES	Type of flow	Label	Local IP	Remote IP	PD	Effacer
114	****	2013-07-02 15:19:04	00:02:45	4.3	4.3	0	0	0	91	0	0	0	27	300	0	0	99.6	0	-25	-67	13648	0	0	0	3	65535	16	75	1	1		192.168.200.122	192.168.200.222	G.711 aLaw	

Reset

While the table is displayed, you can click on a new tablet, the table will be then enriches by the new data of the selected phone call.

Icon XLS located in top on the left of the table, makes it possible to carry out an export of the data of the table towards an Excel file.

This guide applies from the release 4.3.0B02

SCOPE OF SIP ANALYSIS

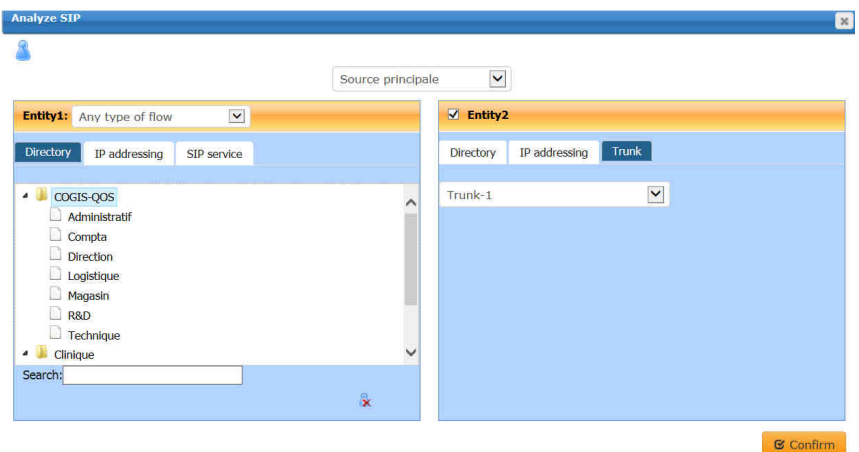
During the operation of the software, either within the monitoring, dashboards or expert module, you will be asked to have to define the scope of analysis.

Then you will get the following screen after clicking on the button "analysis sip."

You must set the criteria defining the perimeter, that is to say the flows that will be analyzed.

Flows are defined by the principle of "flow of Entity1."

You can set the destination flow by defining Entity2.



In the top of the window, select the desired source of acquisition.

Below, on the left, in **Entity 1**, select the type of flow of the entity.

Then below, select the criteria defining the entity, you can choose between:

- Directory: you can select a hierarchy, but also enter an extension number.
- IP Addressing: select an IP address that you have previously defined as one or more addresses.
- SIP Service: select the SIP service on which frames arrive.

If you want to analyze traffic of the extension 3124, in the directory tab, select the hierarchy where is the extension 3124 and below enter " 3124 ".

If you want to analyze traffic "Entity 1" with another entity, on the right check " Entity2 ."

Then define Entity 2:

- Directory : You can enter an extension number or a number outside the compound, knowing that if you specify for the beginning of the number complete with "% " or " % 014510 " for all calls to " 014510 ".
- IP Addressing: select an IP address that you have previously defined as one or more addresses.
- Trunk: select one ou several trunks that you have previously defined.

For settings of IP mappings and trunks, see "Definition of address ranges".

This guide applies from the release 4.3.0B02

DEFINITION OF ADDRESS RANGES

What are address ranges?

Address ranges are used to define the IP address ranges of terminals and trunks.

These ranges are used:

- By the engine of the software, to determine whether a frame corresponds to an outgoing call, incoming or internal call.
- By the operator, to define the scope of his analysis (see "Scope of SIP analysis"), selecting the traffic of one or more positions, one or more trunks.

Handle address ranges

Access to the management of the ranges

To access to the management of the ranges, from the login page, enter your user name and password, then click the icon configurator.



When you have access to the configurator, on the left in the **configuration** tab, click on **SIP** and click:

- **Network Addressing** to define terminal addresses ranges,
- **Configuration of trunks** to define address ranges of trunks.




On the right, you get the data grid where the mappings created appear.

SIP - Network addressing		
<input type="checkbox"/>	Label	IP addressing
<input type="checkbox"/>	cogis-lan	Click here


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Create an addressing

In order to create an addressing, click on the button  , a new line appears within the grid of data.

Indicate the title then click on the button located in the column **Network addressing**.


A new window appears, add the ranges by clicking on the button  and inform the addresses of beginning and end in the form of technical addresses (for example 192.168.1.40 with 192.168.1.69). For a range of only one address, seize the same address in the beginning and end.

Then click on the button 

Modify an addressing

In order to modify an addressing:

- To modify the title, click on the title and seize the new one,
- For the other parameters, click on the button concerned and carry out the necessary modifications.

Then click on the button 

Remove an addressing

Within the grid of data, on the left of addressing to be removed, tick the checkbox

and click on the button 

SIP - Network addressing		
<input type="checkbox"/>	Label	IP addressing
<input type="checkbox"/>	cogis-lan	Click here

This guide applies from the release 4.3.0B02

OTHER ELEMENTS OF CONFIGURATION

What are the terminals compatible with QoS of the VoIP?

To know the terminals compatible with QoS of the VoIP, ask COGIS.

How to set terminals?

Ask COGIS.

Consistency between the frames RTCP-XR and directory

After configuration on the level directory or frames, you may observe an inconsistency between the directory and the assignment of the frames to the directory.

In this case, a tool of consistency is available, to make it possible to reallocate the frames to the datasheets directory.

To access to this tool of consistency, within the module of **configuration**, go in **tools**, then within the pane **tools**, then in lower part click on **purge and recalculate**.

On the right, on the line **SIP coherence** :

- Click on the button **period of analysis**, a window is displayed, define the dates/hours of demarcation of the period over which to carry out coherency, then confirm.
- Of return on the line **SIP coherence**, click on the button **to execute immediately**.

Consistency is launched and is carried out in background task, wait 5 minutes.

Emails profiles

The software sends the emails to email profiles (mailing lists) that you must inform.
An email profile can contain up to 10 emails.

To access to the e-mail profiles, within the configurator, in **configuration**, go in **general**, then on **configure email profiles**.

You can handle the profiles with the buttons **Add** and **Delete**.

Give a title to the profile and inform the recipients' email addresses.

After all modifications, click on **Confirm**.

This guide applies from the release 4.3.0B02

Filter SIP tickets

In what purpose ?

Sometimes, SIP tickets received from the terminals can contain mistakes or data that need to be filtered (removed).

Access to the filter

To access to the management of the surveillances, from the page of identification, inform your **username** and your **password**, then click on the icon of the configurator.



When you have access to the configurator, on the left within **configuration**, click on **SIP**, then on **Filter SIP tickets**.



On the right, you obtain the grid of data where the filters created appear.

Create a filter

In order to create a filter, click on the button , a new line appears within the grid of data.

For the fields that you wish to filter, inform the value to be filtered.

Then click on the button



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This guide applies from the release 4.3.0B02

Modify a filter

In order to modify a filter:

- To modify the title, click on the title and seize the new one,
- For the other parameters, click on the element concerned and inform the new value.

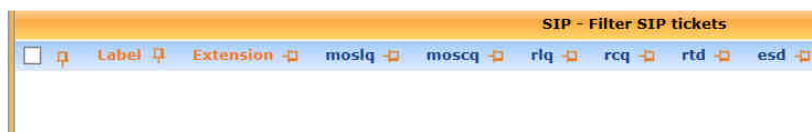
Then click on the button



Remove a filter

Within the grid of data, on the left of the filter to be removed, tick the checkbox

and click on the button



Back Up and archives

Backups and archives are automatically generated, and can be started manually.

User accounts

The software allows to manage various user accounts, each one can be equipped with different rights.

Newspapers

The software has newspapers that bring back the activity of the various processes, some users' actions and the users' entries.

This guide applies from the release 4.3.0B02

Manage and restart processes

Access to the management of processes

To access to the management of processes, from the page of identification, inform your **username** and your **password**, then click on the icon of the configurator.



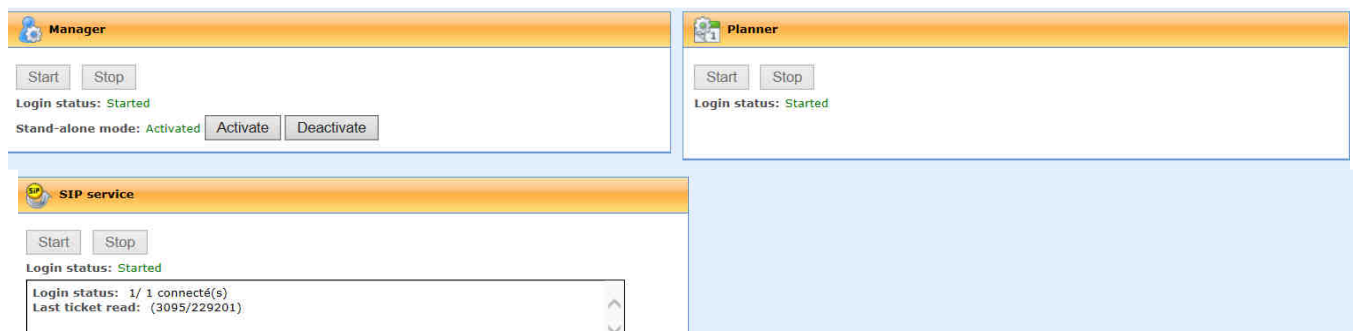
When you are in configurator, on the left within **tools**, click on **tools** then on **Supervision of processes**.

On the right, you obtain as below the grid of data where the processes appear.



What are processes?

They are the various programs integrated within the software. They constitute the engine of the software.



Processes are:

Manager	This process manages the boot of the software and the use of other processes. In case of failure of one of them, it reloads it.
Planner	The planner has in charge the automatic generation of the status reports, the execution of the planned tasks and the management of the email alerts of the SIP surveillance.
SIP service	It is the engine of collecting and processing the frames of QoS of the VoIP.

This guide applies from the release 4.3.0B02

Handle processes

To handle the processes, you must disable the standalone mode.
To do this, within the **Manager** part, click on **Desactivate**.

Once the standalone mode turned-off, the manager is no longer in charge of the monitoring of other processes.

Then, you can start or stop each process.
After clicking a button to start or stop, wait 15 seconds for the display to update.

Once finished, remember to reactivate the standalone mode.
For this in the part **Manager**, click on **Activate**.

Check the acquisition of the frames

To check if the software performs well in acquisition and processing of frames, proceed as follows:

- In the module **configuration**, then **Tools** and **Supervision of processes**.
In the sip service administration, observe what is contained in the last read ticket.
For example: 1710 PUBLISH (176/114600).
From a terminal issuing rtcp-xr frames, make a successful call with more than 5 seconds of conversation and hang up.
Then, you will see change the part of the last ticket.
- Edit the status report **Detailed edition** for the current day, an Excel file is generated, you should observe your call (see Chapter Status reports).

Guide of configuration/use of QoS VoIP

This guide applies from the release 4.3.0B02

Register license

The license can be registered:

- automatically by internet (the server must have access to internet)
- manually by calling COGIS NETWORKS.

To access to the registration of the license, after having identified yourself, go within the **configurator**, in **tools**, then **Help**, click then on **License certificate**.

Inform the form below (serial number, statute book and key) with what is indicated on the license certificate (see example next).

License certificate

Please complete the form using the certificate provided to you:

Serial N° : # 400A - 00 - 70000

Code :

Key : - - - - - -

☐ Upgrade contract / Demo license

Deadline :

Below inform also the name of the company, the operator and

the address e-mail, then click on the button



If internet is functional, the license will be automatically activated.

Then you obtain the following window.

If Not, you obtain the following window:

Then you must activate the license manually by contacting COGIS NETWORKS at +33 1.45.10.31 .08 (Monday to Friday of 9:00 to 12:30 and 13:30 to 18:00, 17:00 on Fridays). Indicate that you're calling for the activation of a license VTP SIP ANALYSER.

You will be asked to give your serial number (that begins with #) which is provided on the license certificate.

Then you will have to provide the provisional code; in exchange a final code will be provided to you. Register the final code within the white field.

Then click on the button:



Document à conserver !

cogis networks

CERTIFICAT DE LICENCE

Ce document est le récépissé de la licence acquise pour le logiciel.

VISUAL TAXE PRO v4

Vous devez impérativement conserver ce document sans limitation de durée. Si vous perdez ce document, la licence ne pourra vous être fournie de nouveau, vous devrez alors acheter une nouvelle licence.

Données constituant la licence logicielle :

Numéro de série	#400A-00-70000
Code (Lettres O majuscules de chiffre zéro)	H D Q I U S J R G O
CM (chiffres seulement)	2 0140 6111000000 3 005 1
Contrat de mise à jour	Aucun contrat

Vous devez saisir ces informations au sein du logiciel, comme indiqué au sein de la documentation d'installation.
Le logiciel va alors générer un code provisoire.
Contactez la société COGIS NETWORKS afin d'obtenir un code définitif.
Date de génération de la licence : 09/01/2010

Document à conserver !

COGIS NETWORKS - 13 avenue Charles de Gaulle - 94401 BOISSEY-LEZ-TOURNAI
Service des licences logicielles - Tél : 01.45.10.31.08 - Fax : 01.45.10.31.11 - logiprog@cogis.com

A propos de VISUAL TAXE PRO ®

VISUAL TAXE PRO ®
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Licence enregistrée
N° série : #400A-00-70000
Version : 4.02A (P03) du 18/07/2011 - 750000 Tickets - 3000 Abonnés annulaire
Options : + 7 Opérateur(s) + 11 Source(s) d'acquisition + Trafic (S + E + L) + Gestion des chambres (3000) + SDA + Sim.

Répertoire d'installation : /usr/local/vtp-manager

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Certificat de licence

N° série : #400A-00-70000

Code provisoire : 1IETI9AHUR

Code licence :

L'enregistrement de la licence par internet est actuellement indisponible. (votre serveur n'a peut être pas accès à internet sur le port TCP/80). Vous pouvez cependant contacter la société **COGIS NETWORKS** par téléphone au **33 (0)1 45 10 31 08** afin d'obtenir votre code licence.

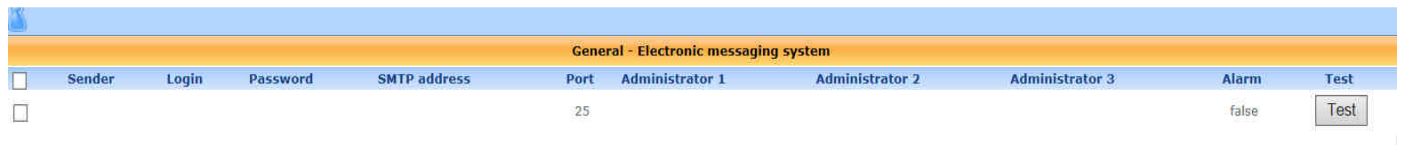
Vous pouvez également recevoir votre licence par mail en prenant soin de nous préciser votre **CODE PROVISOIRE** dans le formulaire de demande d'accès au support technique.
[Support COGIS](#)

This guide applies from the release 4.3.0B02

Set up the SMTP connector

To configure the sending of the e-mails, within **Configuration** then **General**, click on **Electronic messaging system**.

On the right, you obtain this screen :



General - Electronic messaging system										
<input type="checkbox"/>	Sender	Login	Password	SMTP address	Port	Administrator 1	Administrator 2	Administrator 3	Alarm	Test
<input type="checkbox"/>					25				false	Test

To send e-mails, the software uses your SMTP relay.
Contact your network administrator to know the settings and to open flow TCP/25 of the software towards the SMTP relay.

Settings to indicate are:

Field	Settings
Sender	E-mail address attributed to the software. The address may be fictitious, but according to the security of your SMTP relay, it may must be real.
Login (optional)	Optional, depends on the security of your SMTP relay.
Password (optional)	Optional, depends on the security of your SMTP relay.
SMTP Address	IP address or name of the SMTP relay.
Port	TCP port of listening of the SMTP relay (by default TCP/25.
Administrateur1	1st administrator's email address (he will receive logbooks)
Administrateur2 (optional)	2nd administrator's email address (he will receive logbooks)
Administrateur3 (optional)	3rd administrator's email address (he will receive logbooks)
Test	Click this button to send a test email to verify that your configuration is correct.

GLOSSARY OF QoS

QoS Indicators

MOS

MOS

Mean opinion score of quality of conversation.
Rating from 0 to 5.

Transit

Definition

Annuler Transit or delay is a critical parameter strongly influencing the QoS of voice over IP. It represents the time that will put an average IP packet containing a voice sample to cross the infrastructure between two parties.

Jitter

Definition

Jitter or variation of transit time or phase jitter, is the consequence of the fact that all packets containing voice samples will not cross the network at the same speed. This creates a voice distortion or a hash.

Loss

Definition

Transmission of voice per packet is based on the RTP protocol .
This allows to transmit the IP voice packets by reconstituting the information even if the transport layer changes the order of packets.
It uses sequence numbers and relies on UDP.
The real-time latency constraints mentioned above make it unnecessary retransmission of lost packets : RTP even broadcast a datagram arrive too late to be of any use in the process of reconstruction of the voice.
In Voice over IP, lost data is not retransmitted. These losses are due to VoIP data network congestion , that release of packets throughout the network , or excessive jitter will cause discharges packet jitter buffers in the receiver , they cannot receive all the packets arrived in late .
A regular weak loss is less annoying than spaced but higher peaks loss. Indeed human hearing is accustomed to a constant average quality but bear little contrast and sudden degradation of QoS .
The rate of loss in VoIP is typically a few percent or tenths of a percent.

This guide applies from the release 4.3.0B02

Signal

Definition

Measurement audio signal.
Expressed in decibels.

QoS Sub-Indicators

Indicator	Initials	Signification
MOS	MOSLQ	Mean opinion score listening quality
	MOSCQ	Mean opinion score for conversational quality
R Factor	RLQ	R factor of listening quality
	RCQ	R factor of quality conversation
Transit	RTD	Roundtrip time (milliseconds)
	ESD	End system delay (milliseconds)
	SOWD	Delay time
	IAJ	Inter arrival jitter
	UPDATE	Mean absolute jitter
Jitter	JBR	Adjustment of the jitter buffer (milliseconds) speed
	JBN	Buffer within the nominal jitter (milliseconds)
	JBM	Current maximum jitter (milliseconds)
	JBX	Absolute maximum jitter (milliseconds)
	JBA	Adaptability jitter buffer
Loss	NLR	Loss due to network (%)
	JDR	Loss due to jitter (%)
	BLD	Burst density (%)
	GLD	Difference density (%)
	GD	Spread duration (milliseconds)
	GMIN	The deficit threshold
	DATA BASE	Burst duration (milliseconds)
Signal	SL	Signal level (decibels)
	NL	Noise level (decibels)
	RERL	Echo return loss (dB)