WEB GUI 5.3 MANUAL

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Introduction

VoIP monitor is partly open source (sniffer) and partly commercial (GUI/Codecs) VoIP monitoring solution for SIP protocol. The main purpose is to identify SIP call on network and analyses quality of call, record the call to disk (with voice play) and store CDR records to database. This manual covers the WEB GUI part.

Key Features

- Comprehensive search filters - IP, telephone numbers, qualitative parameters (loss/delay/MOS), find all CDR legs
- charts showing call quality and other metrics
- Download PCAP, WAV and online listening to calls via built-in flash player
- SIP REGISTRATION diagnostic tools
- selective voice (RTP) recording
- WEB and Email Report generator
- Alert generator based on various criteria
- Grouping feature based on IP addresses, last SIP response codes, codecs
- Email and IP groups for easy filtering or alerting
- Live calls overview with national/international filter
- User management allowing define users which can see only part of calls based on IP or telephone numbers.
- Listen to call directly from WEB GUI
- Download PCAP or WAV file
- Detailed SIP protocol overview with detail SIP packet (wireshark style)
- More features are planned like billing and alerting based on billing, and much more.

Glossary

Packet loss

Packet loss occurs when one or more packets of data travelling across a computer network fail to reach their destination. Packet loss is distinguished as one of the three main error types encountered in digital communications. Packet loss can be caused by a number of factors including signal degradation over the network medium due to multi-path fading, packet drop because of channel congestion, corrupted packets rejected in-transit, faulty networking hardware, faulty network drivers or normal routing routines.

VoIPmonitor loss

VoIPmonitor detects packet loss and stores loss distribution to 10 loss intervals so it is able to find larger consecutive losses. Thats mainly because you can have two calls with same 2% average packet loss but the first call has random loss distribution and a second call has some “holes” containing larger row of packet losses which is perceived much worse than random loss.

Packet delay variation PDV

In computer networking, packet delay variation (PDV) is the difference in end-to-end one-way delay between selected packets in a flow with any lost packets being ignored. The effect is sometimes referred to as jitter, although the definition is an imprecise fit.

The term PDV is defined in ITU-T Recommendation Y.1540, Internet protocol data communication service - IP packet transfer and availability performance parameters, section 6.2. In computer networking, although not in electronics, usage of the term jitter may cause confusion. From RFC 3393 (section 1.1). In this document, the meaning of jitter will be always same as PDV.

The delay is specified from the start of the packet being transmitted at the source to the end of the packet being received at the destination. A component of the delay which does not vary from packet to packet can be ignored, hence if the packet sizes are the same and packets always take the same time to be processed at the destination then the packet arrival time at the destination could be used instead of the time the end of the packet is received.

For interactive real-time applications, e.g., VoIP, PDV can be a serious issue and hence VoIP transmissions may need Quality of Service-enabled networks to provide a high-quality channel.

The effects of PDV in multimedia streams can be removed by a properly sized jitter buffer at the receiver, which may only cause a detectable delay before the start of media playback.

VoIPmonitor Packet delay variation

VoIPmonitor compares each RTP packet if the delay differs to optimal value (for most cases the delay between two RTP packets are 20ms). If the delay is higher than 50ms it will be counted to one of PDV intervals which is stored for each RPT direction in cdr table. There are those PDV intervals: 50 – 70ms, 70 – 90ms, 90 – 120ms, 120 – 150ms, 150-200ms, > 300ms.

The main advantage over traditional standard jitter metric value is that you can search calls for specific delays characteristics.

Jitter buffer

Jitter buffers or de-jitter buffers are used to counter PDV (jitter) introduced by queuing in packet switched networks so that a continuous playout of audio (or video) transmitted over the network can be ensured. The maximum jitter that can be countered by a de-jitter buffer is equal to the buffering delay introduced before starting the play-out of the mediastream. In the context of packet-switched networks, the term packet delay variation is often preferred over jitter.

Some systems use sophisticated delay-optimal de-jitter buffers that are capable of adapting the buffering delay to changing network jitter characteristics. These are known as adaptive de-jitter buffers and the adaptation logic is based on the jitter estimates computed from the arrival characteristics of the media packets. Adaptive de-jittering involves introducing discontinuities in the media play-out, which may appear offensive to the listener or viewer. Adaptive de-jittering is usually carried out for audio play-outs that feature a VAD/DTX encoded audio, that allows the lengths of the silence periods to be adjusted, thus minimizing the perceptual impact of the adaptation.

MOS score

Mean opinion score (MOS) is a test that has been used for decades in telephony networks to obtain the human user's view of the quality of the network. Historically, and implied by the word Opinion in its name, MOS was a subjective measurement where listeners would sit in a "quiet room" and score call quality as they perceived it; per ITU-T recommendation P.800, "The talker should be seated in a quiet room with volume between 30 and 120 m3 and a reverberation time less than 500 ms (preferably in the range 200-300 ms). The room noise level must be below 30 dBA with no dominant peaks in the spectrum." Measuring Voice over IP (VoIP) is more objective, and is instead a calculation based on performance of the IP network over which it is carried. The calculation, which is defined in the ITU-T PESQ P.862 standard. Like most standards, the implementation is somewhat open to interpretation by the equipment or software manufacturer. Moreover, due to technological progress of phone manufacturers, a calculated MOS of 3.9 in a VoIP network may actually sound better than the formerly subjective score of > 4.0.

In multimedia (audio, voice telephony, or video) especially when codecs are used to compress the bandwidth requirement (for example, of a digitized voice connection from the standard 64 kilobit/second PCM modulation), the MOS provides a numerical indication of the perceived quality from the users' perspective of received media after compression and/or transmission. The MOS is expressed as a single number in the range 1 to 5, where 1 is lowest perceived audio quality, and 5 is the highest perceived audio quality measurement.

MOS tests for voice are specified by ITU-T recommendation P.800

The MOS is generated by averaging the results of a set of standard, subjective tests where a number of listeners rate the heard audio quality of test sentences read aloud by both male and female speakers over the communications medium being tested. A listener is required to give each sentence a rating using the following rating scheme:

Table: MOS rating scheme

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

The MOS is the arithmetic mean of all the individual scores, and can range from 1 (worst) to 5 (best).

Compressor/decompressor (codec) systems and digital signal processing (DSP) are commonly used in voice communications, and can be configured to conserve bandwidth, but there is a trade-off between voice quality and bandwidth conservation. The best codecs provide the most bandwidth conservation while producing the least degradation of voice quality.

Bandwidth can be measured quantitatively, but voice quality requires human interpretation, although estimates of voice quality can be made by automatic test systems.

As an example, the following are mean opinion scores for one implementation of different codecs

Table: MOS for different codecs

Codec	Data rate [kbit/s]	MOS
G.711 (ISDN)	64	4.1
iLBC	15.2	4.14
AMR	12.2	4.14
G.729	8	3.92
G.723.1 r63	6.3	3.9
GSM EFR	12.2	3.8
G.726 ADPCM	32	3.85
G.729a	8	3.7
G.723.1 r53	5.3	3.65
G.728	16	3.61
GSM FR	12.2	3.5

VoIPmonitor MOS prediction

VoIPmonitor transforms PDV and Packet loss into MOS score according to ITU-T E-model which means that the MOS does not represent audio signal but network parameters. Because relation of PDV and MOS score depends on jitterbuffer implementation voipmonitor implements three MOS score

MOS F1 – fixed jitterbuffer simulator up to 50 ms buffer
MOS F2 – fixed jitterbuffer simulator up to 200 ms buffer
MOS adapt – adaptive jitterbuffer simulator up to 500ms buffer

VoIPmonitor assumes that the call uses G711 codec with maximum MOS score 4.5. Thats why calls does not have “right” subjective 4.1. The reason is that you can easily filters all calls for the same MOS score regardless on used codec. If you want to have real MOS score for G.729 – there is option in sniffer (check /etc/voipmonitor.conf).

The MOS score should not be taken as a definitive value. You have to check delay/loss distribution and other paratmeters. This value is just for quick filtering of potentially bad calls.

Post Dial Delay (PDD)

Post Dial Delay (PDD) is experienced by the originating customer as the time from the sending of the final dialled digit to the point at which they hear ring tone or other in-band information. Where the originating network is required to play an announcement before completing the call then this definition of PDD excludes the duration of such announcements.

RTCP

The RTP Control Protocol (RTCP) is a sister protocol of the Real-time Transport Protocol (RTP). Its basic functionality and packet structure is defined in the RTP specification RFC 3550 superseding its original standardization in 1996 (RFC 1889).

RTCP provides out-of-band statistics and control information for an RTP flow. It partners RTP in the delivery and packaging of multimedia data, but does not transport any media streams itself. Typically RTP will be sent on an even-numbered UDP port, with RTCP messages being sent over the next higher odd-numbered port. The primary function of RTCP is to provide feedback on the quality of service (QoS) in media distribution by periodically sending statistics information to participants in a streaming multimedia session.

RTCP gathers statistics for a media connection and information such as transmitted octet and packet counts, lost packet counts, jitter, and round-trip delay time. An application may use this information to control quality of service parameters, perhaps by limiting flow, or using a different codec.

VoIPmonitor (version >= 5) is able to parse and store RTCP statistics. For each call RTCP jitter, fraction loss and total loss is saved for each direction.

Installation

This section describes WEB GUI installation for Debian and Redhat derivates. VoIPmonitor standard version is encoded with ionCube (tools to protect software written using the PHP programming language from being viewed, changed, and run on unlicensed computers). To be able to decode ionCube encoded PHP script – the ionCube zend extension has to be loaded to PHP. The ionCube loader extension is available for Linux, FreeBSD, OpenBSD, OS X, Solaris and Windows and the installation is described in this section.

This installation procedure assumes that you have running voipmonitor sniffer which covers sniffer manual downloadable from http://www.voipmonitor.org/download – Sniffer manual

Starting from WEB GUI ver. 4 build 215 there are installation instructions directly in the web browser.

Prerequisite packages for Debian/Ubuntu

```
apt-get install php5-gd php5-mysql php5 php5-cli apache2 libapache2-
mod-php5 tshark mtr
```

Prerequisite packages for Centos/Redhat

```
yum install httpd wireshark php php-gd php-mysql php-mbstring mtr
php-process
```

Package installation

Download the latest VoIPmonitor GUI from http://www.voipmonitor.org/download and place it to /var/www on debian/ubuntu or to /var/www/html on centos/redhat

```
cd /var/www (or /var/www/html)
tar xzf voipmonitor-gui*.tar.gz
rm voipmonitor-gui*.tar.gz
mv voipmonitor-gui-5.0* voipmonitor
```

Download license key.php from http://www.voipmonitor.org/download for later use.

Point your web browser to http://yourserver/voipmonitor and follow the installation/configuration instructions.

Cleaning old data crontab

In the GUI folder there is script which cleans old pcap files until 10% of disk free space.

Debian/Ubuntu

```
/etc/cron.daily/voipmonitor

#!/bin/bash
/usr/bin/php  /var/www/voipmonitor/php/run.php
removeOldCaptureFiles  -s 10% -f
```

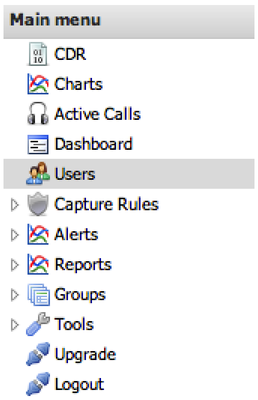
Centos/Redhat/Fedora

```
/etc/cron.daily/voipmonitor

#!/bin/bash
/usr/bin/php  /var/www/html/voipmonitor/php/run.php
removeOldCaptureFiles  -s 10% -f
```

Cleaningn old CDR records

This is completely up to the user how and when the old CDR should be deleted. Recommended way is to wipe out older records in regular intervals to not overgrow database. Deleting is very expensive operation on large MySQL tables.

User management

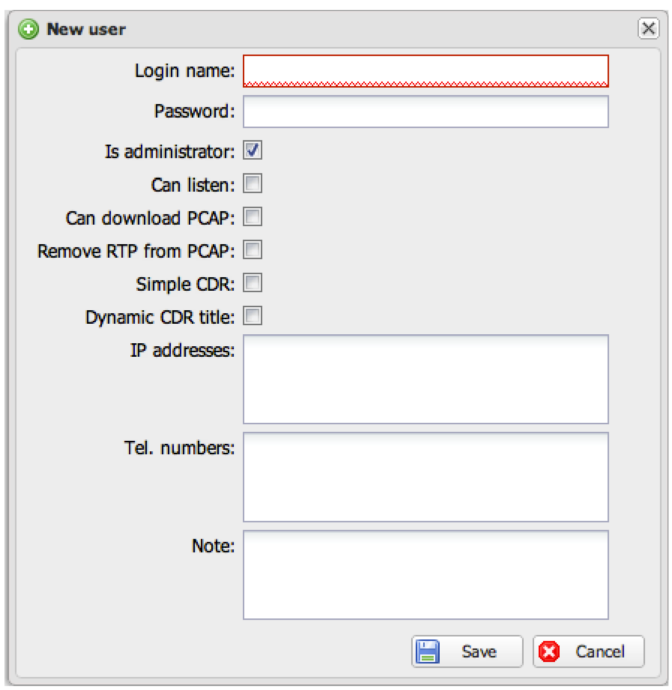


VoIP monitor allows define multiple user accounts with different rights. **If no user is defined** user admin with password admin is active. Once there is one user defined, the admin/admin account no longer exists so be careful that you create full admin user before you logout from admin/admin. If your session expires in web browser (which depends on PHP default settings which is around 2 hours) the WEB GUI will prompt you for relogin. Users are saved in database table users. If you cannot login delete all users

```
echo "delete from users" | mysql voipmonitor
```

Creating new user

Click on New user button and fill the New user form. Then click on Save.



Login name + Password are used for login to the WEB GUI.

Is administrator – has rights to create/delete/modify users and to all features

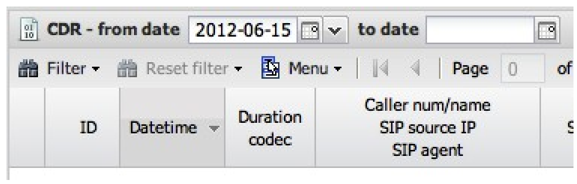
Can listen – user can listen to WAV or can download WAV files

Can download PCAP – user can download PCAP file

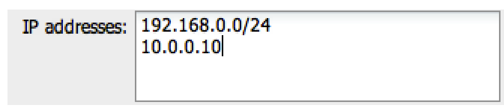
Remove RTP from PCAP – if user will click on PCAP download the RTP stream will be removed from the PCAP file (but still remains on disk)

Simple CDR – user will view only simple CDR layout without QoS metrics. This is usefull for users which wants to see basic CDR overview and wants to listen to calls (callcenters, etc.).

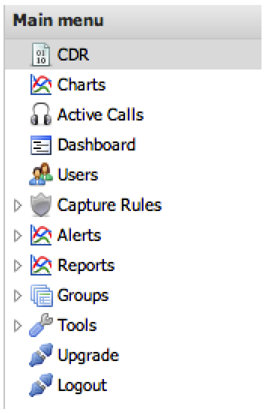
Dynamic CDR title – this will show date filter in CDR title.



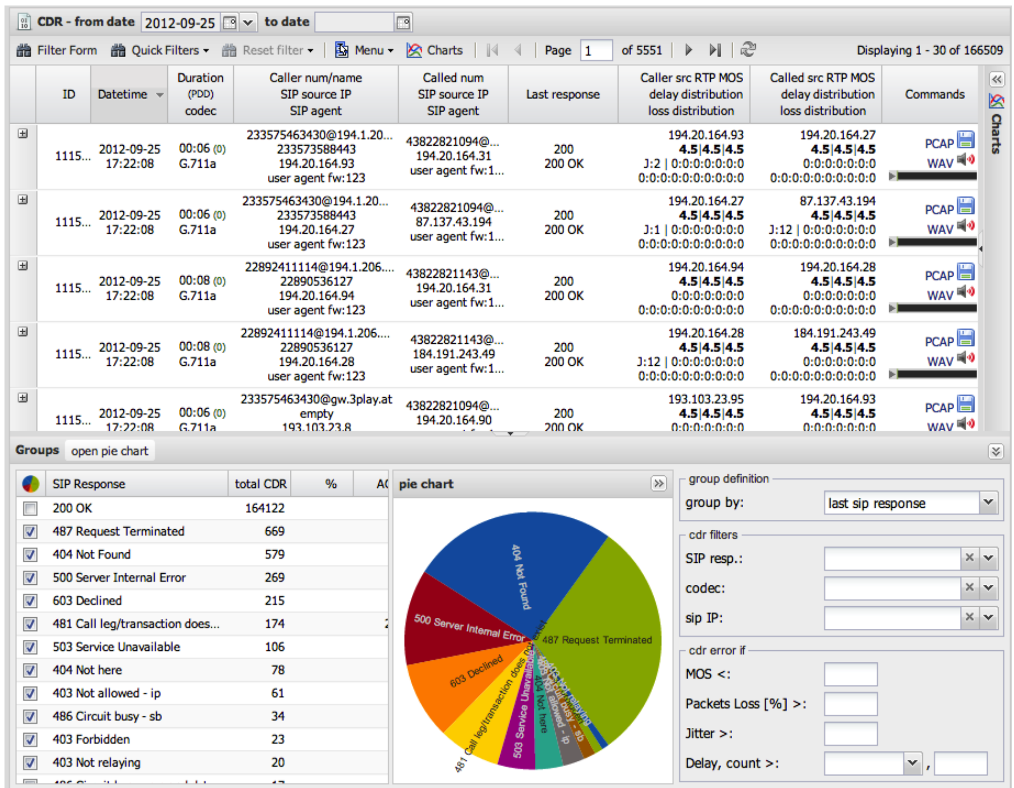
IP addresses – list of allowed IP addresses or IP networks to see by user. This option is usefull to restrict users to view only certain CDR. The list of IP addresses has to be delimited by [enter]. Example:



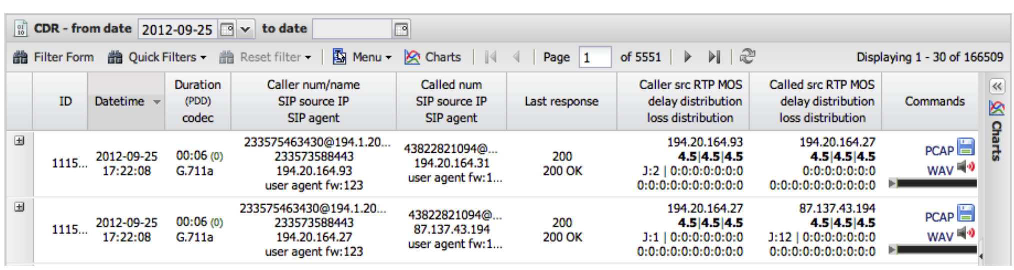
Tel. Numbers – list of allowed telephone numbers. To restric user to telephone prefixes use '%' - for example all numbers started with 222%



CDR shows all saved and finished calls in database cdr table. CDR main window is divided to CDR list and Dashboard at the bottom. Dashboard can be resized or hidden 



CDR list



Title bar

CDR list starts with Title bar where you can quickly filter calls based on date range. (please note that this date range will not be active if you disabled it in user preferences).

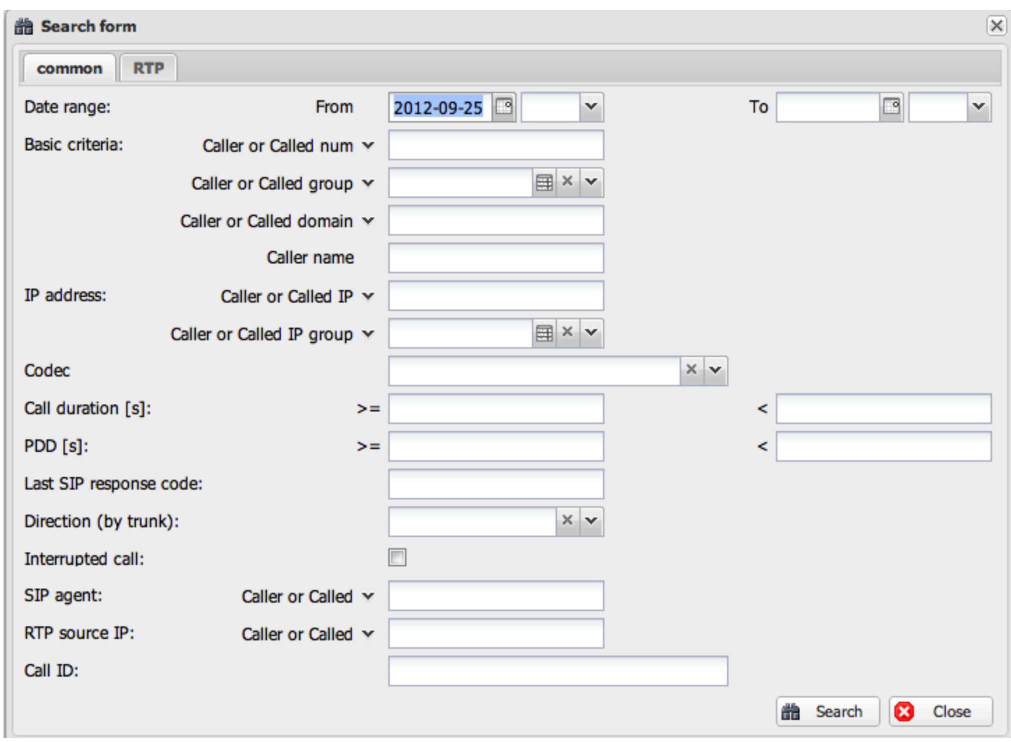
Button panel

Below the CDR title is Button panel where you can list through CDR pages and access filters and other features:

Filter Form button

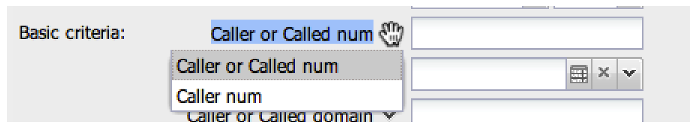
clicking on Filter Form button shows advanced Search form with two tabs on top - “common” and “RTP”.

Common tab



Date range filters CDR based on Date and/or Hour/Minute criteria.

Caller called num or name or domain can be filtered for specific number/string or for specific prefix “222%” or specific suffix “%222”. (please note that searching for suffix uses reversed column with index and is as fast as searching for prefix). You can also search only for Caller or only for Called number clicking on arrow and select Caller num.



Caller or called group can be used for searching for specific list of numbers. You can manage groups directly by clicking on the group icon.

IP address – use single IP address or specific network like 192.168.0.0/24. CDR is filtered by SIP IP signalization.

Call duration filters by specific duration interval (total call length including ringing).

PDD – Search Post Dial Delay range.

Last SIP response code filters by SIP status codes (like 483, 503, 603 etc). To find all 4XX responses use 4% syntax.

Direction (by trunk) – filter calls by direction IN, OUT or Internal. To distinct direction you need to create IP group and set trunk checkbox on it. Interal calls are all which does not match the IP list of all Trunk IP groups.

Interrupted call checkbox finds all interrupted calls which are those without BYE or confirmation to BYE.

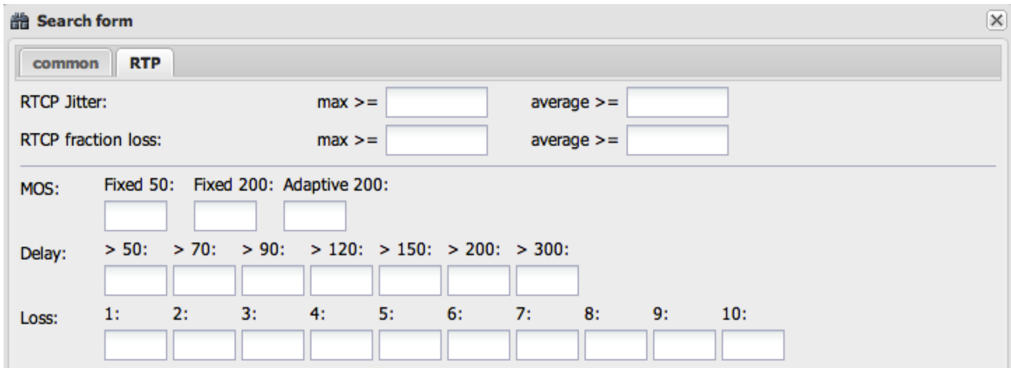
RTP source IP – filters calls by RTP source IP addresses instead of SIP IP addresses.

SIP agent filters SIP agent header. This header usually carries phone manufacturer/firmware version.

Call ID filters SIP Call-ID hedear which is unique string. This string also names pcap files.

RTP tab

RTP tab is used to filter calls by RTP metrics.



RTCP Jitter – filters calls by the worst RTCP jitter value of both directions either by its MAX value or average value.

RTCP fraction loss – filters calls by the worst RTCP fraction loss value of both directions either by its MAX value or average value.

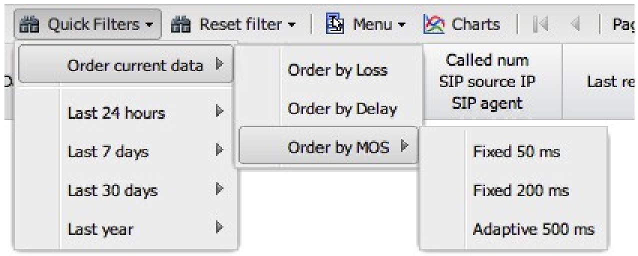
MOS – filters all calls which have MOS lower than entered value.

Delay – Search calls by PDV intervals. To find really bad calls use PDV intervals >120 for at least 10 occurrences.

Loss – Search calls by number of consecutive loss. Number 1 represents number of single packet occurrences, number 2 is number of two consecutive lost packets, ..., number 10 is number of more than 10 consecutive lost packets.

Quick filters

Is used to find the worst calls by Loss, Delay or combination of that two (MOS).



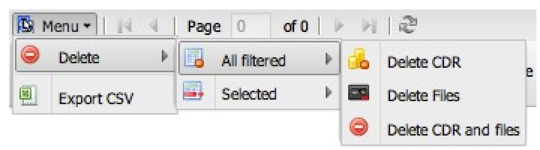
Oder current data will order current filtered CDR by Loss, Delay or MOS score.

Last 24 hours, 7 days, 30 days and yeat will order by worst Loss, Delay or MOS score. Take in mind that ordering milions of CDR (month or year) can take a lot of time.

Reset filter button

This button resets searching criteria to default values.

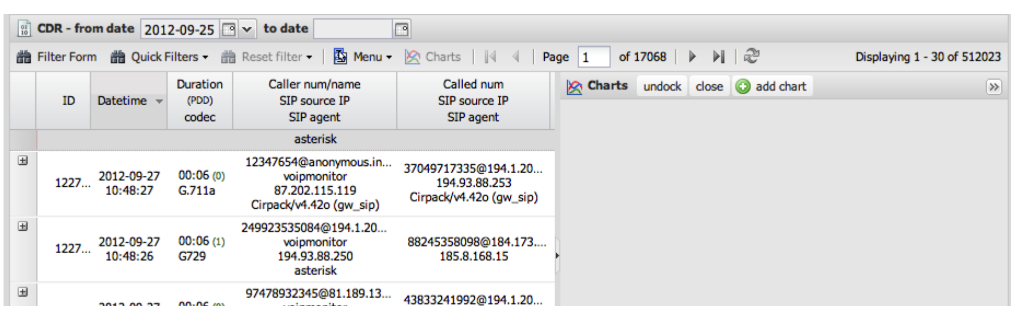
Menu button

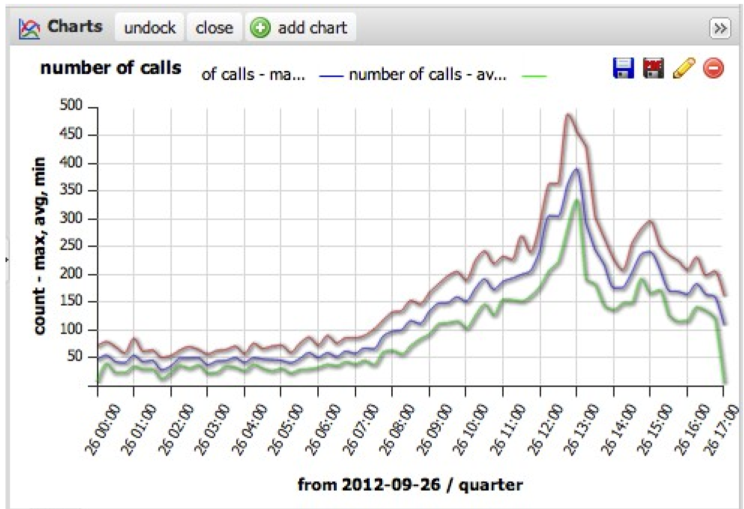


Under Menu button is Delete and Export CSV. Delete allows delete CDR records and files either for all current filtered CDRs or only for selected filters (you can select several CDR by holding CTRL+mouse click).

Charts button

Activates small window with charts which is used to add charts to current data.





undock

Undock button detaches chart window to float window

close

Close button hides charts window.

add chart

Add chart buttons shows chart form for adding graph.

Saves graph as SVG

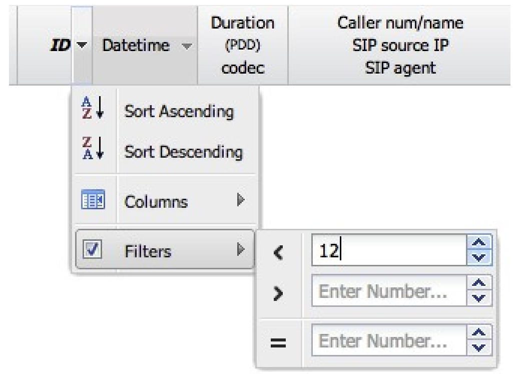
Modifies created graph

Removes graph

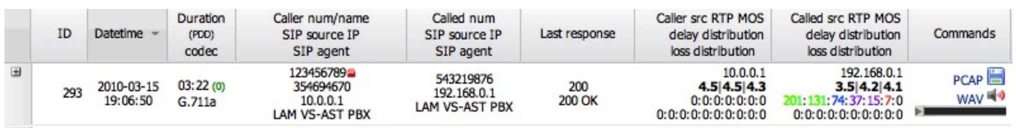
Charts are described in detail in chapter “Charts”.

CDR column headers

Column headers contains quick filters and some of it are sortable. Some of the columns is also possible to hide or reorder by dragging it.



CDR row



CDR row contains this columns:

ID – it is unique number increasing by one for each new CDR.

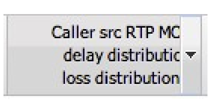
Datetime – is start of the call

Duration (PDD)/Codec – shows Duratino of call, PDD and used codec.

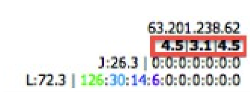
Call num/name, SIP source IP, SIP agent shows information identifying caller.

Called num, SIP destination IP, SIP agent shows information identifying callee.

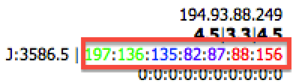
Last response – shows number and full text last SIP response. For connected calls it shows 200 OK.



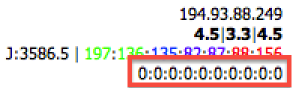
Column Caller/Called RTP shows source IP address of Caller/Called RTP stream. The IP address represents SOURCE IP of Caller or Called RTP stream.



MOS Score row shows MOS score for three type of jitterbuffer – fixed 50, fixed 200 and adaptive 500.

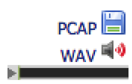


Delay distribution shows all PDV intervals colored accordingly – left number is 50 – 70ms interval and has green color. The most right number is PDV interval >300ms and has red color.



Loss distribution shows all loss intervals colored accordingly – left number is one consecutive loss occurrences and has green color. The most right number is more than 10 consecutive loss occurrences and has red color.

RTCP part (if RTCP packet was captured) shows MAX RTP jitter (J:26.3) and MAX RTCP fraction loss (L:72.3). RTCP in Caller column shows how called side sees the stream.



Commands shows two links and one flash based WAV player. PCAP will download PCAP file and WAV link will download audio file.

Play button starts playing directly in web browser (flash plugin has to be installed).

CDR detail

Clicking on [+] shows full detail of the CDR with extended informations.

Detail area starts with tabs – first one is summary, next is SIP: history followed by Legs by CID and Legs by header.

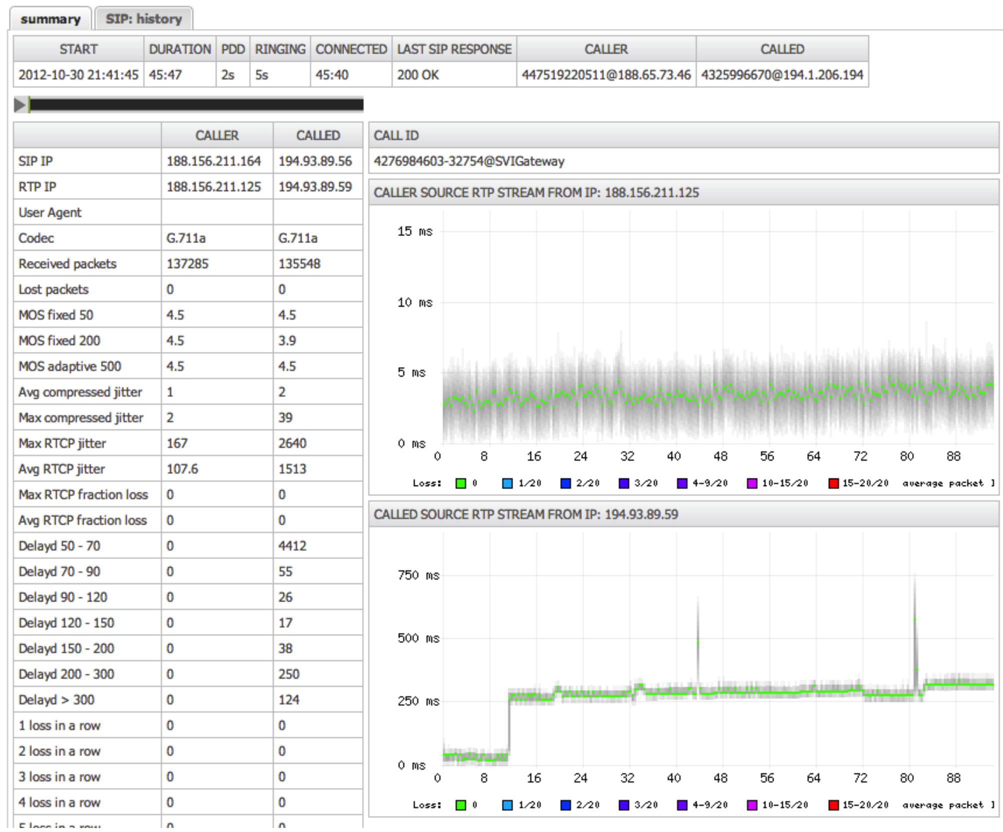
Summary

The first table shows SIP signalization information like call start, duration, PDD time, ringing time and connected time, last SIP reposne, caller and called information. Under this table bigger WAV player is shown.

The next table shows RTP statistics, PDV intervals and loss intervals. Most of values are self-explanatory except those

Avg compressed jitter represents PDV where number 1 is no jitter (or very little). Higher number represents higher jitter. *Max compressed jitter* shows maximum jitter during the call. This value is described in RTP RFC.

The last table Shows SIP messages chronologically. Each SIP message is clickable where new WINDOW appears with full packet information with all protocols Ethernet – IP – UDP – SIP/RTP.

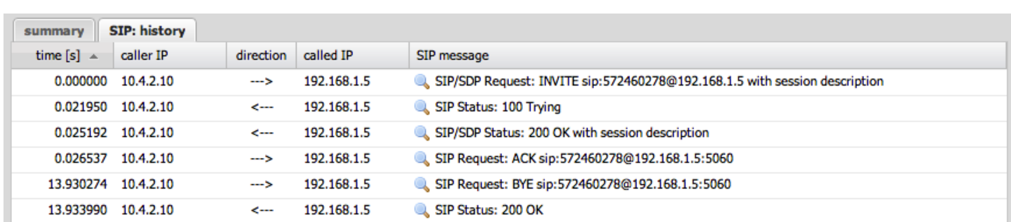


START	DURATION	PDD	RINGING	CONNECTED	LAST SIP RESPONSE	CALLER	CALLED
2012-10-30 21:41:45	45:47	2s	5s	45:40	200 OK	447519220511@188.65.73.46	4325996670@194.1.206.194

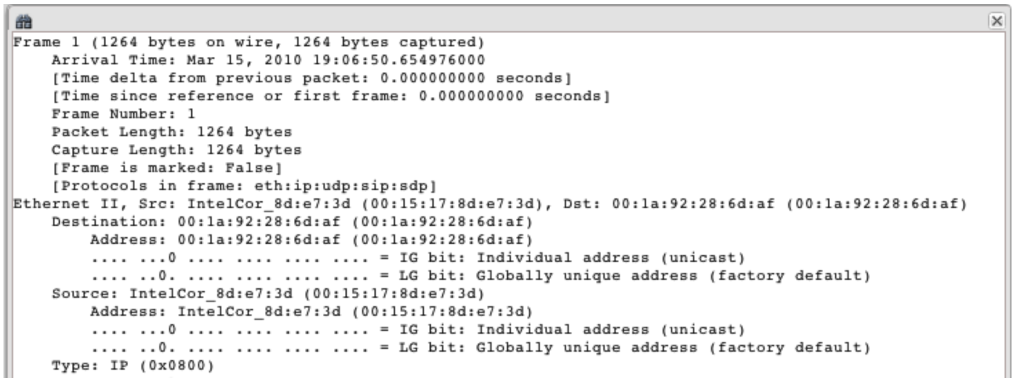
	CALLER	CALLED
SIP IP	188.156.211.164	194.93.89.56
RTP IP	188.156.211.125	194.93.89.59
User Agent		
Codec	G.711a	G.711a
Received packets	137285	135548
Lost packets	0	0
MOS fixed 50	4.5	4.5
MOS fixed 200	4.5	3.9
MOS adaptive 500	4.5	4.5
Avg compressed jitter	1	2
Max compressed jitter	2	39
Max RTCP jitter	167	2640
Avg RTCP jitter	107.6	1513
Max RTCP fraction loss	0	0
Avg RTCP fraction loss	0	0
Delayd 50 - 70	0	4412
Delayd 70 - 90	0	55
Delayd 90 - 120	0	26
Delayd 120 - 150	0	17
Delayd 150 - 200	0	38
Delayd 200 - 300	0	250
Delayd > 300	0	124
1 loss in a row	0	0
2 loss in a row	0	0
3 loss in a row	0	0
4 loss in a row	0	0

SIP: history

SIP history tab shows SIP packets chronologically. Each message can be viewed in detail by clicking on magnifier icon.

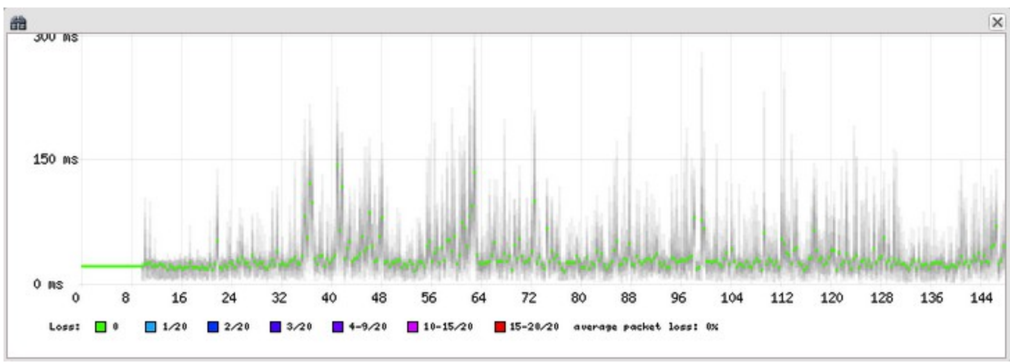


time [s]	caller IP	direction	called IP	SIP message
0.000000	10.4.2.10	--->	192.168.1.5	SIP/SDP Request: INVITE sip:572460278@192.168.1.5 with session description
0.021950	10.4.2.10	<---	192.168.1.5	SIP Status: 100 Trying
0.025192	10.4.2.10	<---	192.168.1.5	SIP/SDP Status: 200 OK with session description
0.026537	10.4.2.10	--->	192.168.1.5	SIP Request: ACK sip:572460278@192.168.1.5:5060
13.930274	10.4.2.10	--->	192.168.1.5	SIP Request: BYE sip:572460278@192.168.1.5:5060
13.933990	10.4.2.10	<---	192.168.1.5	SIP Status: 200 OK



```
Frame 1 (1264 bytes on wire, 1264 bytes captured)
    Arrival Time: Mar 15, 2010 19:06:50.654976000
    [Time delta from previous packet: 0.000000000 seconds]
    [Time since reference or first frame: 0.000000000 seconds]
    Frame Number: 1
    Packet Length: 1264 bytes
    Capture Length: 1264 bytes
    [Frame is marked: False]
    [Protocols in frame: eth:ip:udp:sip:sdp]
Ethernet II, Src: IntelCor_8d:e7:3d (00:15:17:8d:e7:3d), Dst: 00:1a:92:28:6d:af (00:1a:92:28:6d:af)
    Destination: 00:1a:92:28:6d:af (00:1a:92:28:6d:af)
        Address: 00:1a:92:28:6d:af (00:1a:92:28:6d:af)
        .... ...0 .... .... .... .... = IG bit: Individual address (unicast)
        .... ..0. .... .... .... .... = LG bit: Globally unique address (factory default)
    Source: IntelCor_8d:e7:3d (00:15:17:8d:e7:3d)
        Address: IntelCor_8d:e7:3d (00:15:17:8d:e7:3d)
        .... ...0 .... .... .... .... = IG bit: Individual address (unicast)
        .... ..0. .... .... .... .... = LG bit: Globally unique address (factory default)
    Type: IP (0x0800)
```

Graph section shows detailed delay and loss distribution. Clicking on the graph will open new window with the graph.



- Each vertical tick represents 20 received packets
- The color dot represents PDV median from 20 received packets
- Gray lines represents PDV variation, max and low values. If helps optically how the PDV spreads over the call.
- Color of a dot represents packet loss. The legend is below the graph. Green dot is 0% packet loss. Red dot is more then 19 packet loss.

Legs by CID

Legs by CID is grid of CDR which matches by caller id number and start of the call + - 5 seconds (default). This interval can be adjusted in the tool bar. Purpose of this grid is quick way of finding all calls which might belongs to the same call – for example if call is routed through asterisk and asterisk calls to hunt group VoIPmonitor creates for each leg CDR – Incoming leg and outgoing legs. To find both legs click on one of the call.

Legs by header

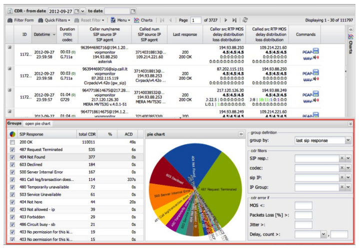
Legs by header is grid of CDR which matches by match_header which can be any SIP header (set it in /etc/voipmonitor.conf). Typical is in-reply-to header.

Here is example of 6 CDR which is connected by match_header and callid header.

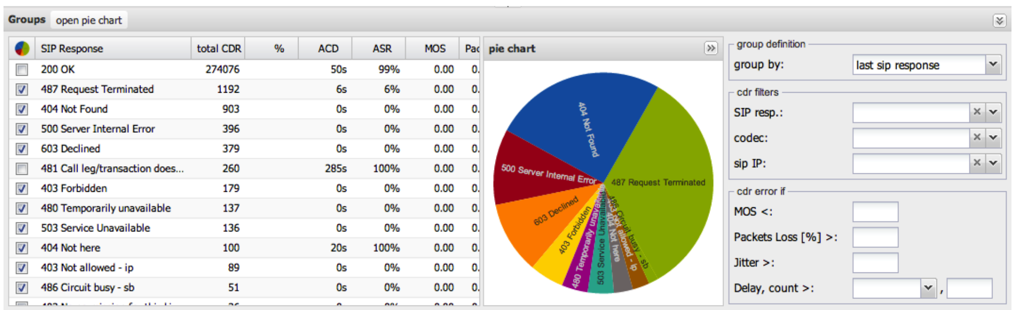
```
+-----------------------------------------------+-----------------------------------------+
| call-id                                       | match_header                            |
+-----------------------------------------------+-----------------------------------------+
| 26c4480371f54e170ede2dcf1caf1599@213.242.88.118 | TP4D2OC4KRDL5F6V3LDC3L6DEM@81.201.83.45 |
| 4b07692c119f643119997e5833e9fe83@213.242.88.118 | TP4D2OC4KRDL5F6V3LDC3L6DEM@81.201.83.45 |
| 5b800a5c5e86d12833fe44ec4eac4ae8@213.242.88.118 | TP4D2OC4KRDL5F6V3LDC3L6DEM@81.201.83.45 |
| TP4D2OC4KRDL5F6V3LDC3L6DEM@81.201.83.45       | NULL                                    |
| 5b800a5c5e86d12833fe44ec4eac4ae8@213.242.88.118 | NULL                                    |
| 5b800a5c5e86d12833fe44ec4eac4ae8@213.242.88.118 | NULL                                    |
+-----------------------------------------------+-----------------------------------------+
```

Legs by CID and Legs by header is also able to search in remote MySQL databases. To enable this feature create sensors in Tools -> Sensors section.

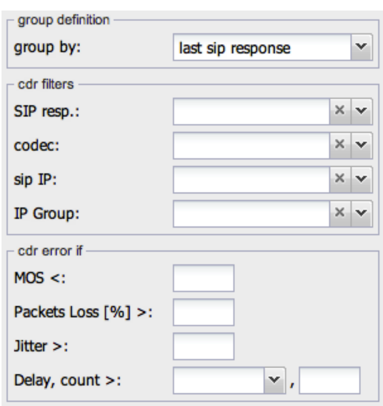
CDR groups panel



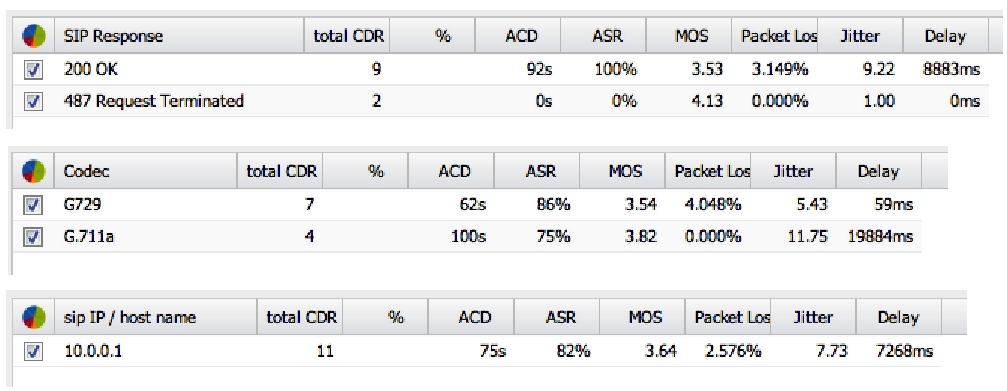
CDR groups panel is devided into three sections. The left section shows grid of data releated to choosen group. The middle section represents grid data in PIE chart. The right section controls which data and how should be presented. The CDR groups panel is tight with the upper CDR view list – for example clicking on 200 OK SIP responses will filter all calls based on 200 OK reponess.



SIP Response	total CDR	%	ACD	ASR	MOS
200 OK	274076		50s	99%	0.00
487 Request Terminated	1192		6s	6%	0.00
404 Not Found	903		0s	0%	0.00
500 Server Internal Error	396		0s	0%	0.00
603 Declined	379		0s	0%	0.00
481 Call leg/transaction does...	260		285s	100%	0.00
403 Forbidden	179		0s	0%	0.00
480 Temporarily unavailable	137		0s	0%	0.00
503 Service Unavailable	136		0s	0%	0.00
404 Not here	100		20s	100%	0.00
403 Not allowed - ip	89		0s	0%	0.00
486 Circuit busy - sb	51		0s	0%	0.00



Group By – choose which groups you would like to see. You can choose last sip response, Codecs, SIP IP or IP group which is group of IP addresses defined in Group main menu.



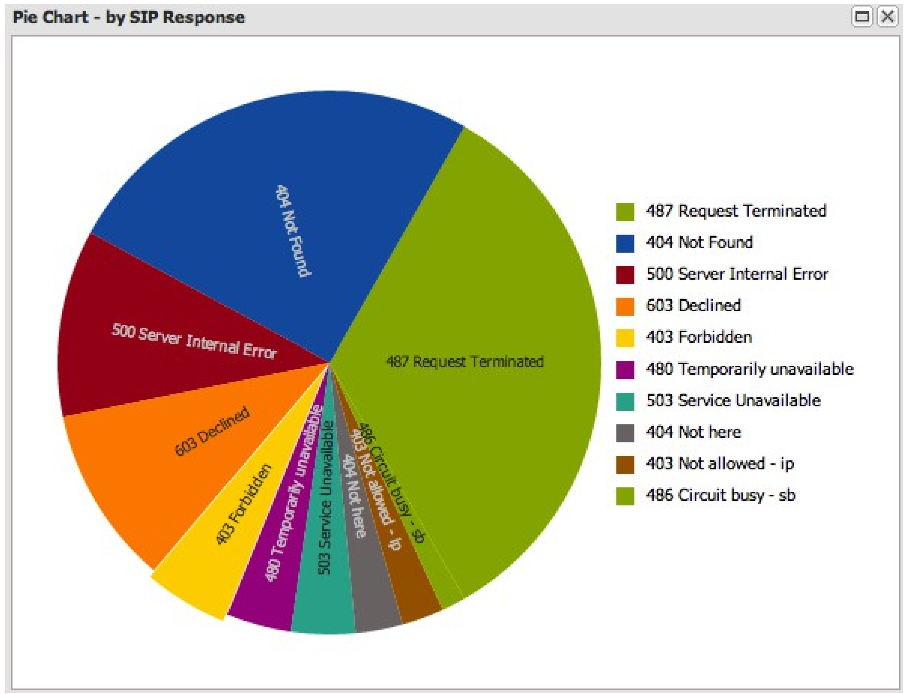
SIP Response	total CDR	%	ACD	ASR	MOS	Packet Los	Jitter	Delay
200 OK	9		92s	100%	3.53	3.149%	9.22	8883ms
487 Request Terminated	2		0s	0%	4.13	0.000%	1.00	0ms

Codec	total CDR	%	ACD	ASR	MOS	Packet Los	Jitter	Delay
G729	7		62s	86%	3.54	4.048%	5.43	59ms
G.711a	4		100s	75%	3.82	0.000%	11.75	19884ms

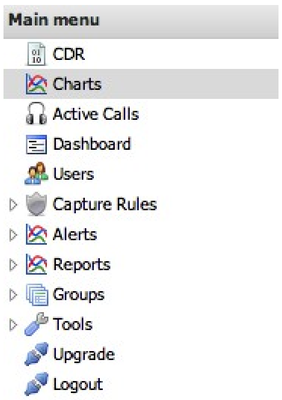
sip IP / host name	total CDR	%	ACD	ASR	MOS	Packet Los	Jitter	Delay
10.0.0.1	11		75s	82%	3.64	2.576%	7.73	7268ms

ACD – The Average Call Duration (ACD) is calculated by taking the sum of billable seconds (billsec) of answered calls and dividing it by the number of these answered calls.
ASR - The Answer-Seizure Ratio (ASR) is calculated by dividing the number of successfully answered calls by the total number of calls attempted, which are known as "seizures". **60-70% i**s considered a very good ASR in the VoIP world.

Clicking on Groups open pie chart new window with pie chart is opened. In that window hiding and showing particular data can be achieved by clicking on it in the right legend. Hovering over the color will show percentual value.



Charts

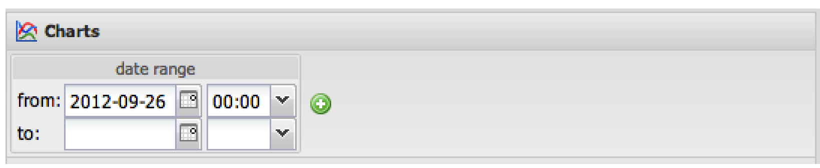


Charts is used to plot various data sources like number of concurrent calls or quality of calls over time. Data sources can be combined to one chart allowing to see correlation of desired data sources like SIP 4XX/5XX responses on ASR.

Charts is also present in CDR window in right mini-window sharing the same functionality with only difference that in CDR section the graph takes data from the current filter.

Add graph

To create new graph click on + button.



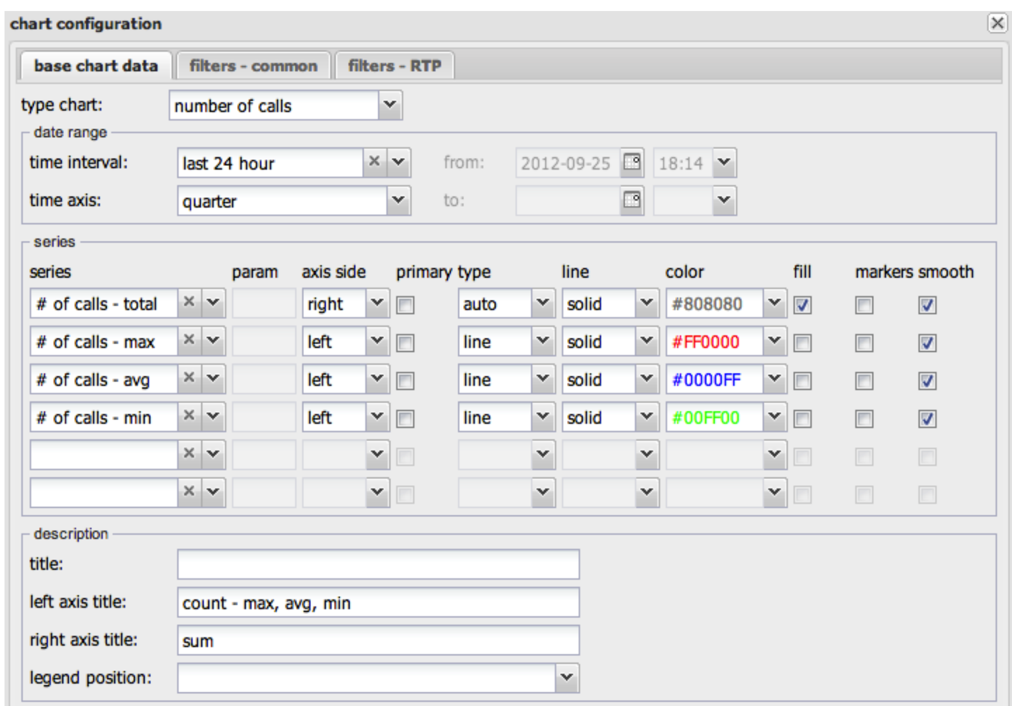
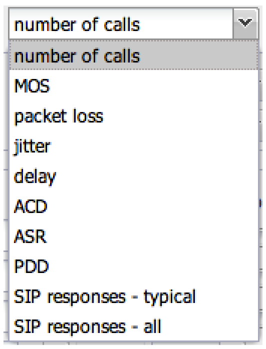


Chart configuration

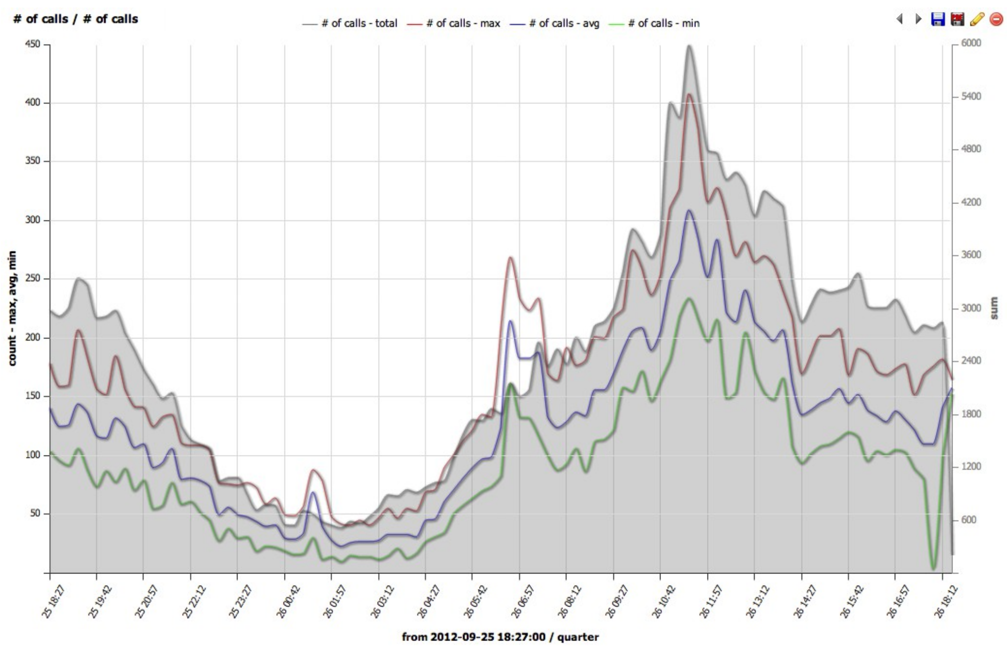
chart configuration contains three tabs at the top – base chart data, filters – common and filters – rtp. Filters tab is used to filter data sources by various criteria – for detailed description please refer to CDR chapter.

Chart configuration panel is divided to three sections. Type chart + interval, series and description.



Type chart field contains predefined chart configurations:

number of calls is graph with 4 datasources – number of total calls made (gray) and number of simultaneous calls – MAX, AVG, MIN.



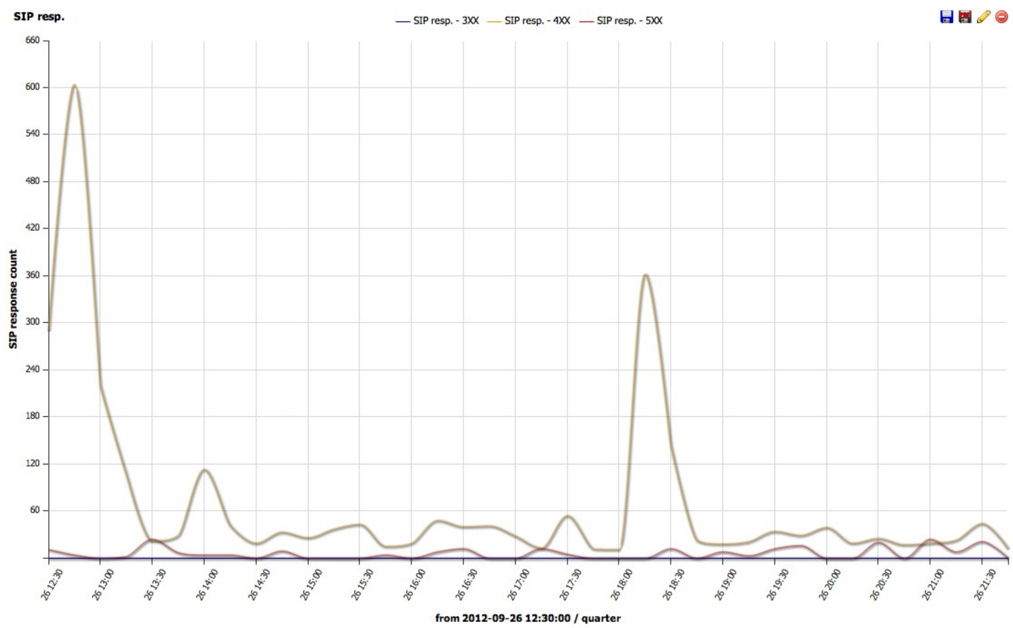
The graph itself is interactive and reacts on some items – top legend (clicking on particular legend hides datasource). Hovering on datasource highlights it and shows local value.

MOS, packet loss, jitter and delay shows RTP statistics.

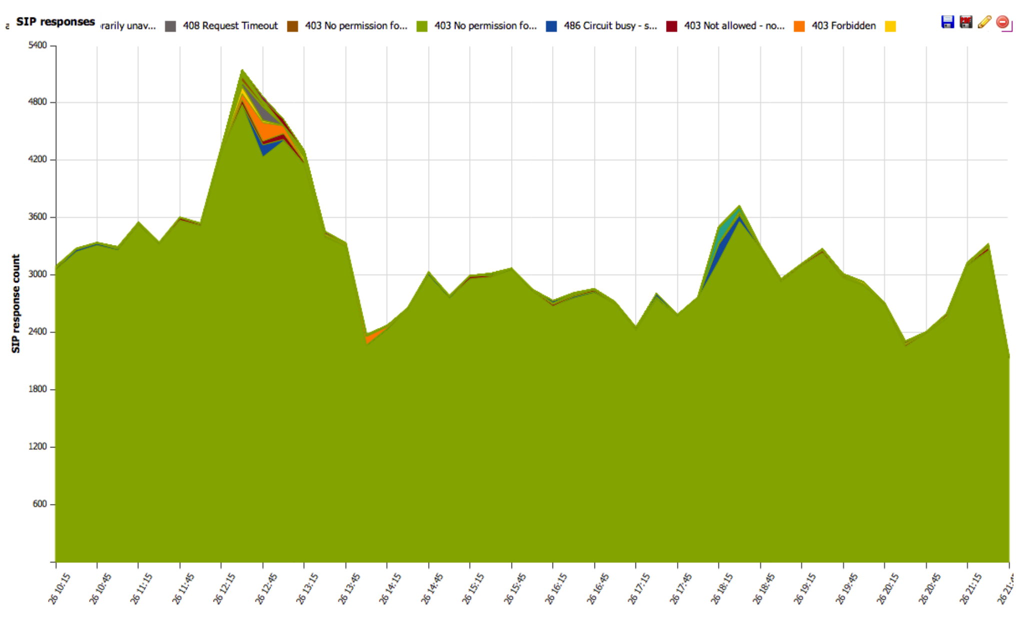
ACD – average call duration, ASR – average seizure ratio

PDD - Post Dial Delay

SIP responses – typical – shows SIP 3XX, 4XX and 5XX distribution

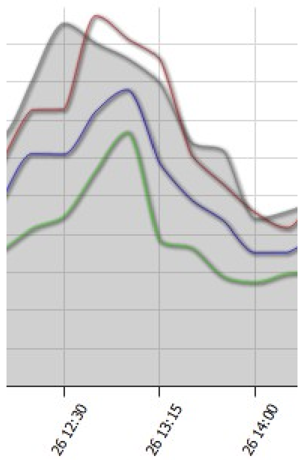


SIP responses – all shows stacked graph of all SIP responses

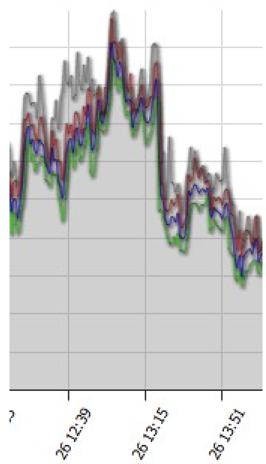


Date range is density on X axe.

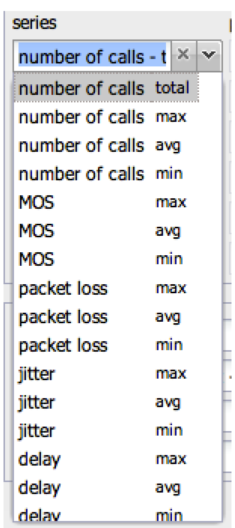
Quarter

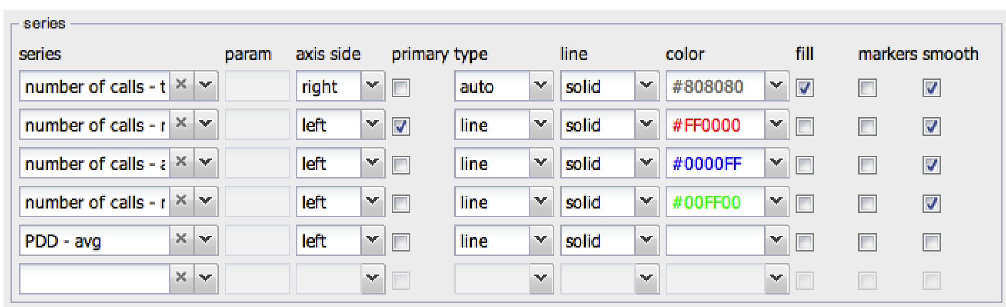


Minutes



Series panel builds graphs.

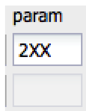




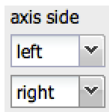
List of series:

number of calls total – total number of created calls
number of calls max/min/avg number of simultaneous calls
MOS max/avg/min
Packet loss max/avg/min
jitter max/avg/min
delay max/avg/min
ACD
ASR
PDD max/avg/min

SIP resp. - custom sip response
SIP resp. [2345]XX – all 2XX-5XX responses
SIP responses – stacked graph



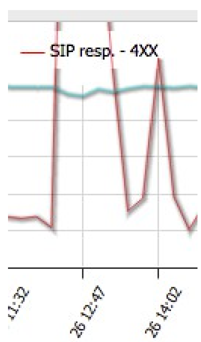
param is used for SIP responses

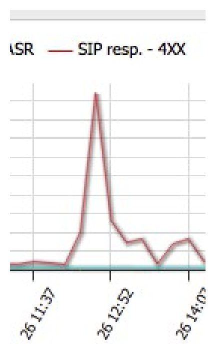


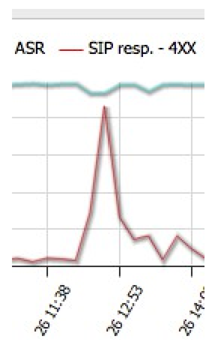
Axis side is used to assign data source to left Y axe or to right Y axe.

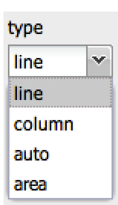
Primary checkbox is used in case where more datasources are drawn on left or right Y. Primary checked datasource fills the whole Y axe and non-primary datasources use that scale. Here is example:

Blue is primary | Red is primary | Blue left, Red right



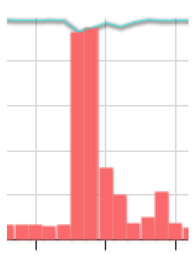


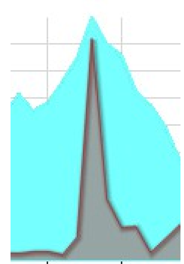


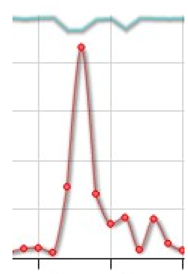


Type of graph

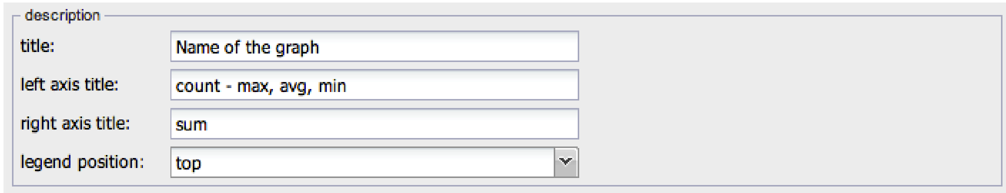
Line + Column | Area + Line(filled) | Smooth line + markers



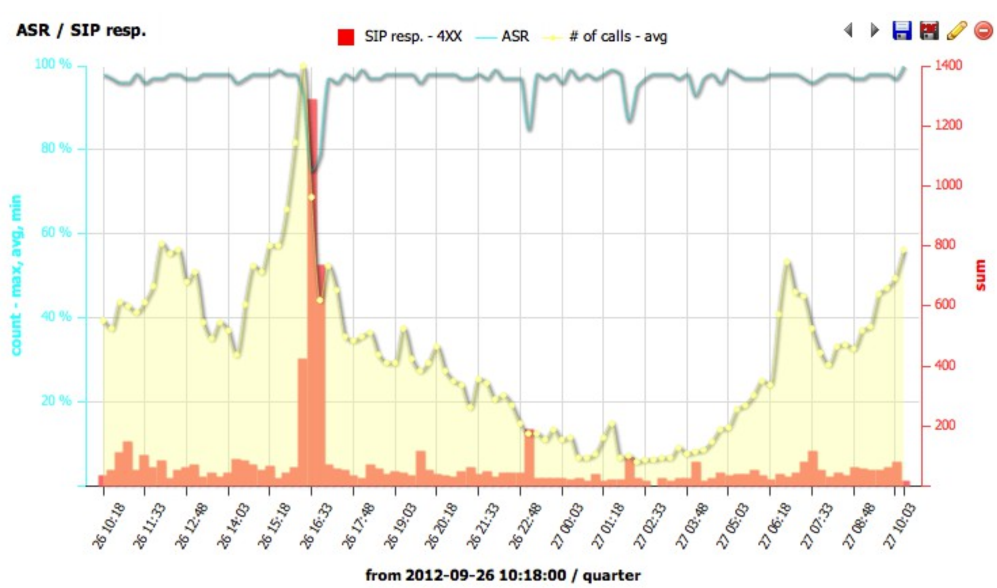




Description names graph, axis and sets legend position (top/left/right/bottom or no legend)

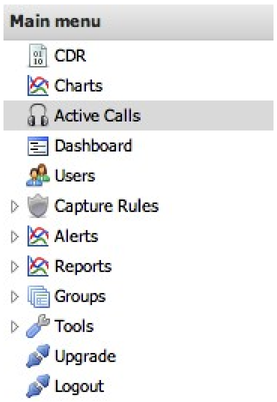


Setting your own title will allow to save created graph for repetitive use. Once the title is filled or changed, save button will appear next to type chart.



Here is example of combined graph – Blue line is ASR, Yellow area is number of simultaneous calls and RED bars are number of 4XX responses. On this graph we see that calls around 16:33 dropped suddenly, ASR dropped and SIP 4XX responses increased. This gap was caused by one faulty device which was restarted immediately.

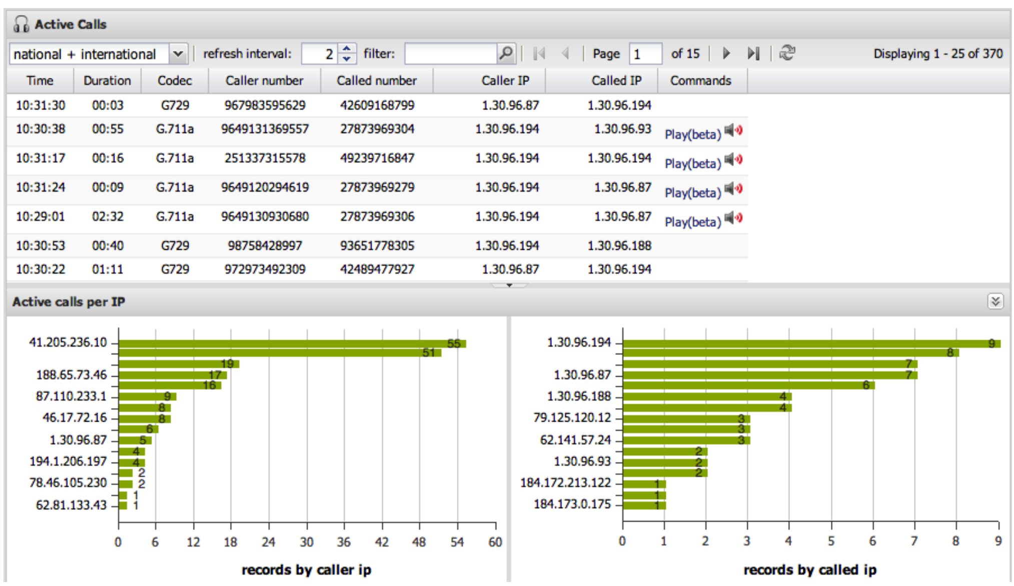
Live calls



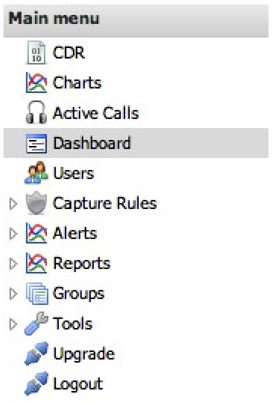
Live calls shows current calls in realtime. It refreshes according to refresh interval (default 2 seconds). Calls can be filtered by national or international by clicking on combo box. Filter is also used to filter by IP, IP prefix or number. Depends on input the filter box adapts to correct search. For example 192.168 will filter all calls with source or destination IP addresses starting with 192.168.0.0/16. Providing only number for example 00 will filter all calls starting with 00.

Bottom graphs shows top most calls by caller IP or called IP.

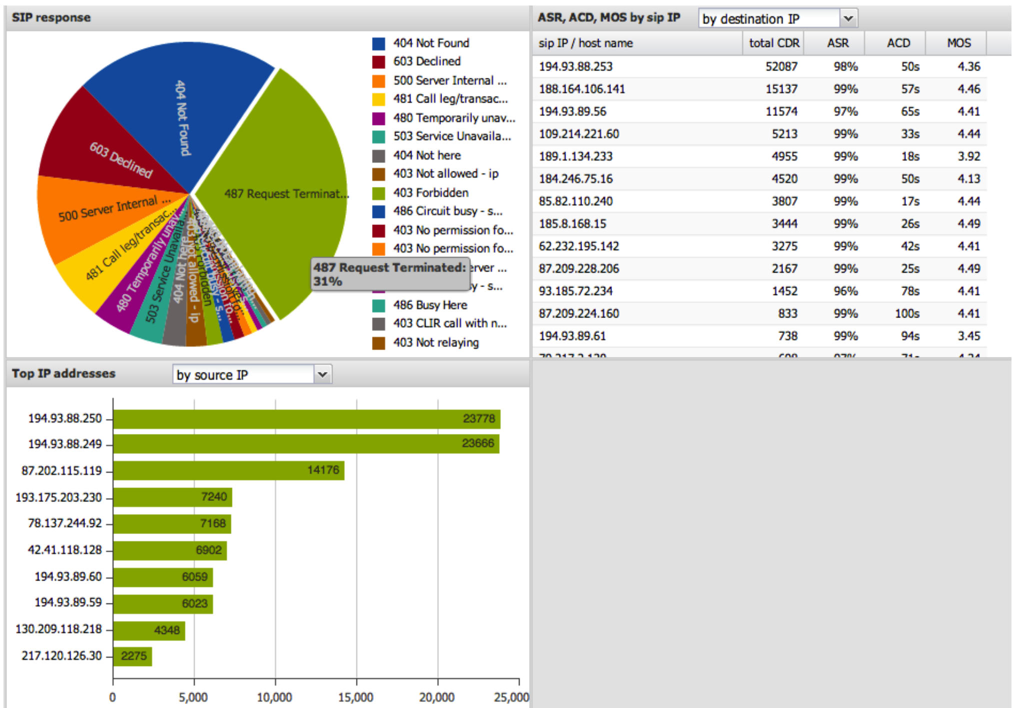
Live calls are fetched from voipmonitor instance through manager TCP port 5029. If calls are not shown please check on the web server if it has access to that port (for example by telnet localhost 5029).



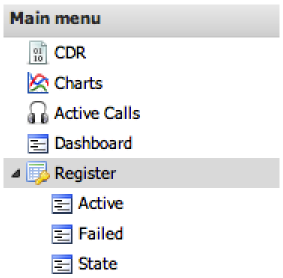
Dashboard



Dashboard currently implements three views for current day – SIP responses, ASR/ACD/MOS and Top IP addresses view ordered by top most source or destination IP addresses. Dashboard will be enhanced in future versions to allow placing custom charts and other various widgets.



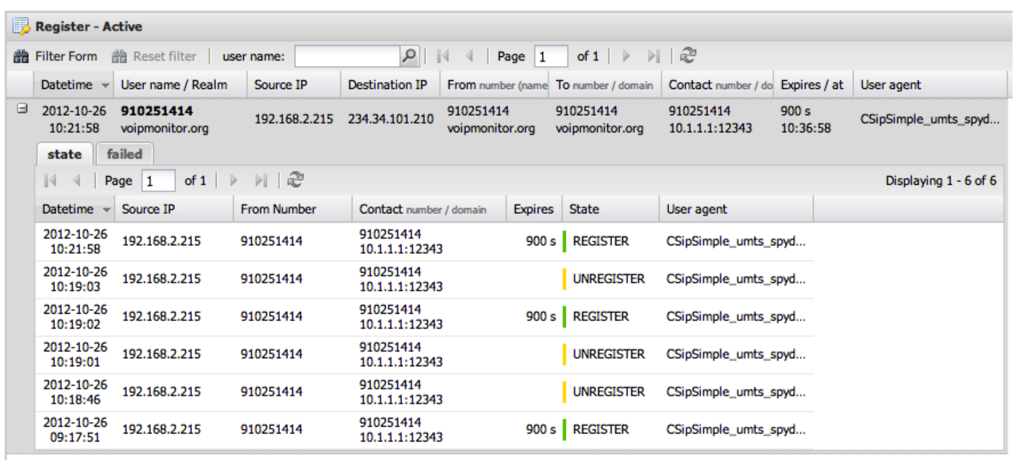
Register



SIP Register section shows three tables - Active registered SIP users, Failed registrations and State changes in SIP registrations. Those tables are filled once you enable sip-register = yes in /etc/voipmonitor.conf

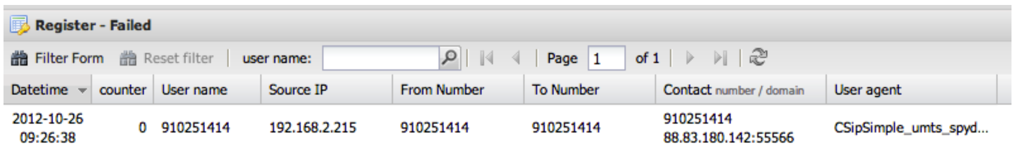
Active table

Active table shows current registered users. On the picture below you can see detail area where is subgrid with state changes and failed registrations from the user name – this is quick filters for particular active user where you can quickly see his history. Once the SIP registration is expired is is not in Active table anymore. Each expired registration is stored in State table.



Failed table

Failed table shows failed SIP registrations. If some device fails to register continuously the counter column is increasing instead of creating new row. If there is 1 hour gap between two failed registrations from the same user – next row is created.

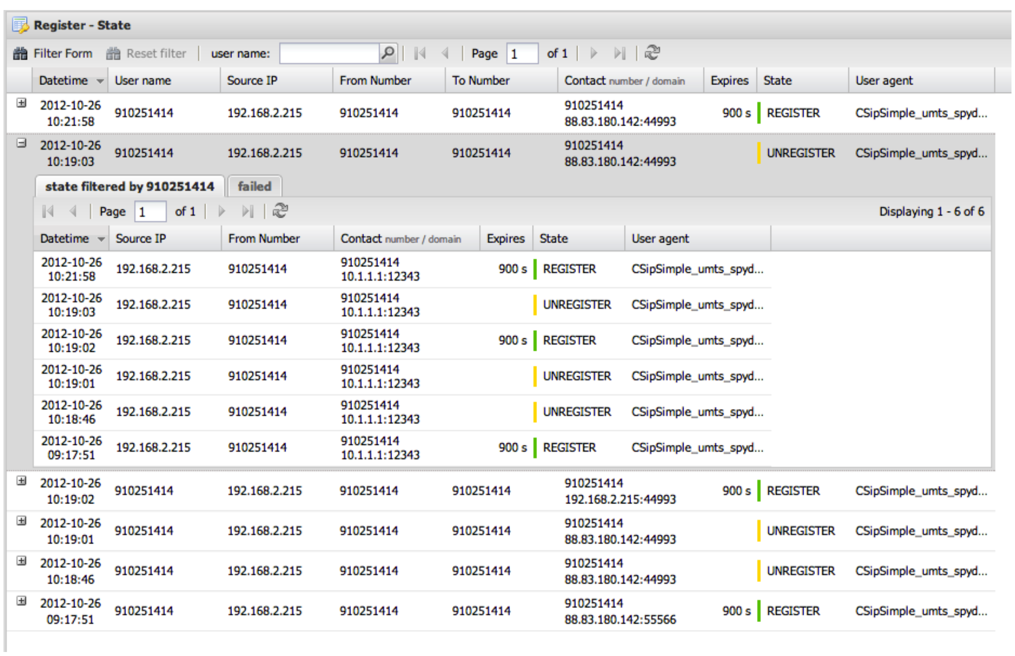

Register - Failed

Filter Form | Reset filter | user name: | Page 1 of 1

Datetime	counter	User name	Source IP	From Number	To Number	Contact number / domain	User agent
2012-10-26 09:26:38	0	910251414	192.168.2.215	910251414	910251414	910251414 88.83.180.142:55566	CSipSimple_umts_spyd...

State table

State table retains registration history where REGISTER, UNREGISTER and EXPIRE is saved. In each state row you can click on detail [+] to show all releated SIP messages to clicked user (and also failed).


Register - State

Filter Form | Reset filter | user name: | Page 1 of 1

	Datetime	User name	Source IP	From Number	To Number	Contact number / domain	Expires	State	User agent
⊞	2012-10-26 10:21:58	910251414	192.168.2.215	910251414	910251414	910251414 88.83.180.142:44993	900 s	REGISTER	CSipSimple_umts_spyd...
⊟	2012-10-26 10:19:03	910251414	192.168.2.215	910251414	910251414	910251414 88.83.180.142:44993		UNREGISTER	CSipSimple_umts_spyd...

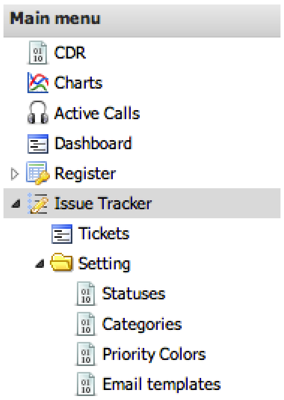
state filtered by 910251414 | failed

Page 1 of 1 | Displaying 1 - 6 of 6

Datetime	Source IP	From Number	Contact number / domain	Expires	State	User agent
2012-10-26 10:21:58	192.168.2.215	910251414	910251414 10.1.1.1:12343	900 s	REGISTER	CSipSimple_umts_spyd...
2012-10-26 10:19:03	192.168.2.215	910251414	910251414 10.1.1.1:12343		UNREGISTER	CSipSimple_umts_spyd...
2012-10-26 10:19:02	192.168.2.215	910251414	910251414 10.1.1.1:12343	900 s	REGISTER	CSipSimple_umts_spyd...
2012-10-26 10:19:01	192.168.2.215	910251414	910251414 10.1.1.1:12343		UNREGISTER	CSipSimple_umts_spyd...
2012-10-26 10:18:46	192.168.2.215	910251414	910251414 10.1.1.1:12343		UNREGISTER	CSipSimple_umts_spyd...
2012-10-26 09:17:51	192.168.2.215	910251414	910251414 10.1.1.1:12343	900 s	REGISTER	CSipSimple_umts_spyd...

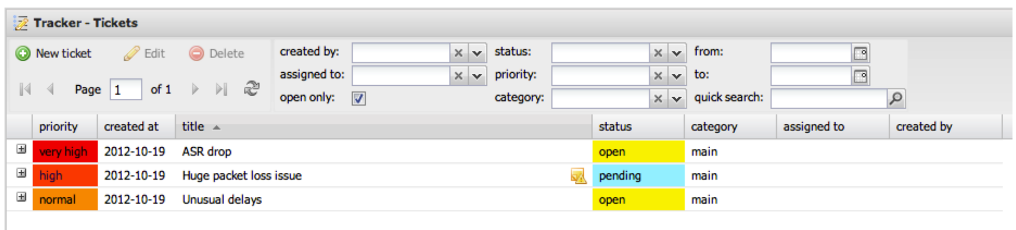
	Datetime	User name	Source IP	From Number	To Number	Contact number / domain	Expires	State	User agent
⊞	2012-10-26 10:19:02	910251414	192.168.2.215	910251414	910251414	910251414 192.168.2.215:44993	900 s	REGISTER	CSipSimple_umts_spyd...
⊞	2012-10-26 10:19:01	910251414	192.168.2.215	910251414	910251414	910251414 88.83.180.142:44993		UNREGISTER	CSipSimple_umts_spyd...
⊞	2012-10-26 10:18:46	910251414	192.168.2.215	910251414	910251414	910251414 88.83.180.142:44993		UNREGISTER	CSipSimple_umts_spyd...
⊞	2012-10-26 09:17:51	910251414	192.168.2.215	910251414	910251414	910251414 88.83.180.142:55566	900 s	REGISTER	CSipSimple_umts_spyd...

Issue tracker



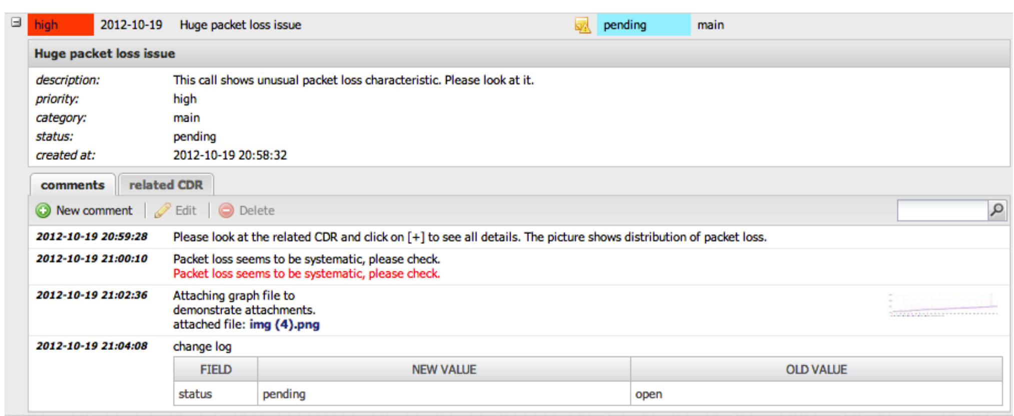
Issue tracker is tool meant mainly for ITSP operators where the operator can create ticket and assign to someone to solve the trouble.

Tickets grid shows by default all not closed tickets. To create new ticket press New ticket button. Ticket can be assigned to different categories and can be assigned to some user. The user is notified by email that new ticket was created and assigned to him. Each changes is sent over email to creator and to assigned or participated users.



Ticket comments

Each ticket has its own history which is shown by clicking on [+]. Ticket can also have file attachments like pictures or any kind of files. Ticket can also has CDR relations which links CDRs directly to ticket (assigning CDR to ticket is done in main CDR section). To see all assigned CDRs click on related CDR tab. Each change in ticket (like closing ticket) is logged in comments.



Setting

Stuses

Defines status of tickets. Status can be open or closed can have own color and name. One of the status can be set as default which is then selected in new created ticket.

Tracker - Statuses

New status | Edit | Delete

status	type	default	description
closed	closed		
open	open	✓	
pending	open		Ticket is beeing processed

Categories

Categories is used to categorize tickets and can have its own text and background color. Default category is main.

Tracker - Categories

New category | Edit | Delete

category	default	description
main	✓	

Priority colors

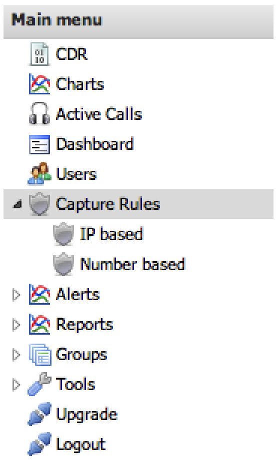
Here you can modify priority colors, name and set default priority.

Tracker - Priority Colors

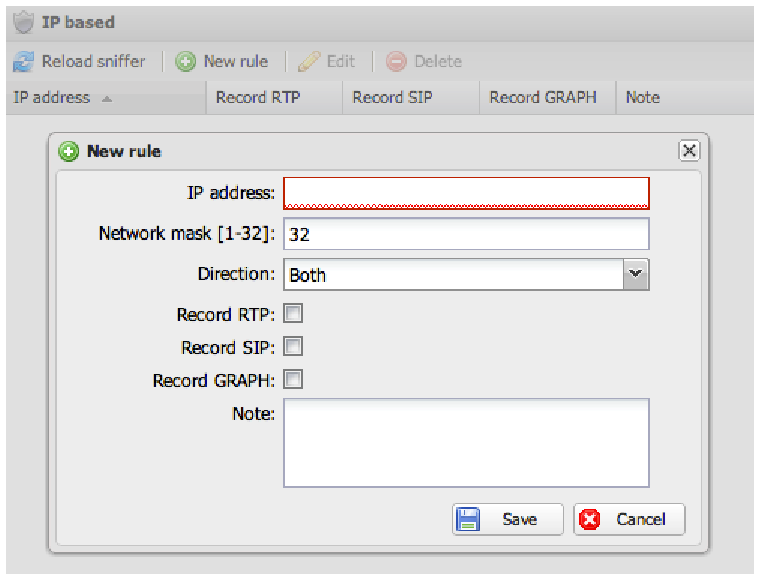
Edit

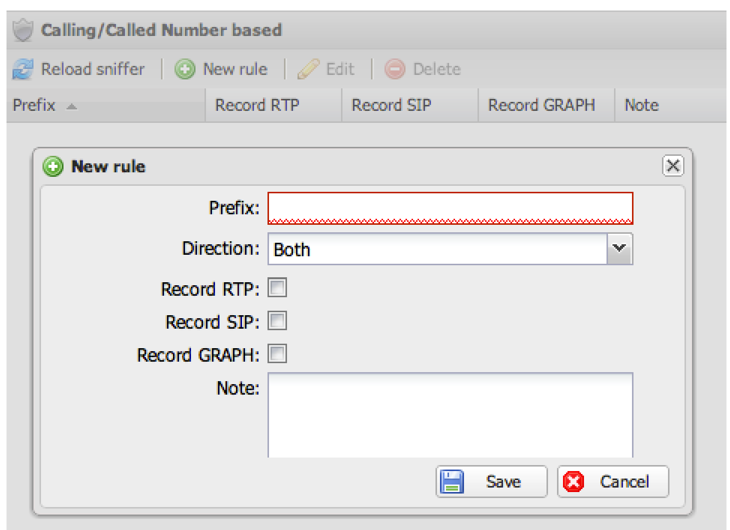
priority	default	description
very low		
low		
normal	✓	
high		
very high		

Capture Rules

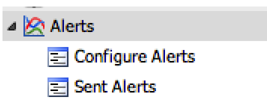


Capture rules allows to capture only certain calls to disk. Typical is to not save complete RTP packets to disk (or only the RTP headers) and allow to capture full RTP packets or Graphs or SIP signalization based on IP or number rules. Sniffer is loading rules on start and allows to reload rules without restarting the service. Reload rules has to be done by clicking on Reload sniffer button.





Alerts



Alerts&Reports contains tools to generate email alerts based on QoS parameters or SIP error conditions. It can also generate daily report or generate ad hoc reports. All generated alerts and reports are saved in history.

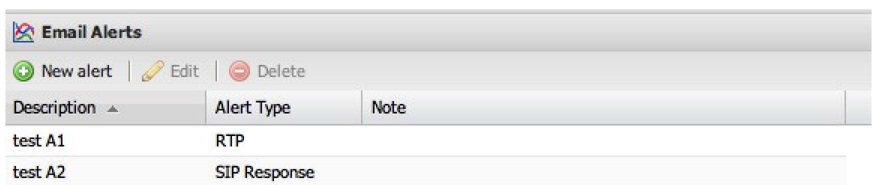
Alerts are processed by PHP script which has to be placed to crontab

```
/etc/cron.d/voipmonitor

01 0 * * * root php /var/www/voipmonitor/php/run.php reports
-r alert@example.com -s

*/5 * * * * root php /var/www/voipmonitor/php/run.php alerts
-r alert@example.com -s
```

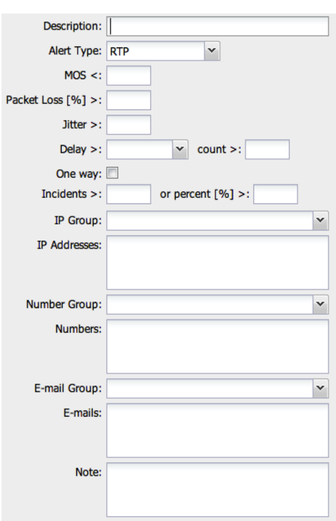
Do not forget to killall -HUP cron (crond)

Configure Alerts

Email alerts triggers alerts based on SIP protocol or RTP QoS metrics.

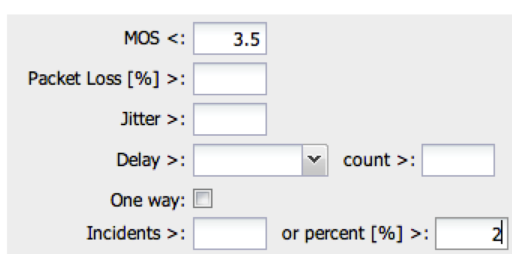


New alert rule

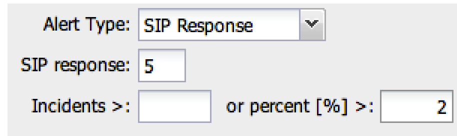


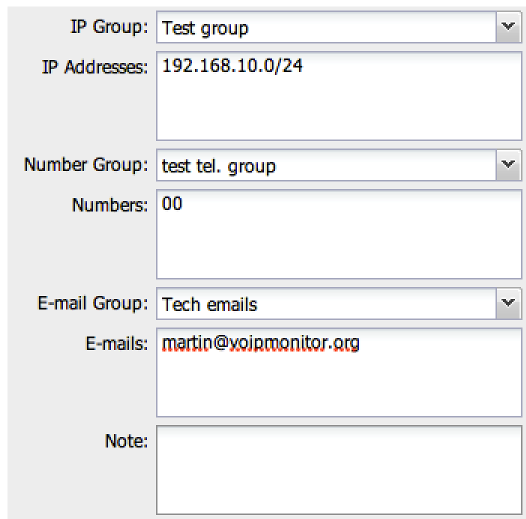
Alert is divided to two types RTP and SIP response. Each of those shares common filters: IP addresses, Numbers and E-mails to which the alert is sent.

Alert type RTP allows to trigger alert based on MOS, Packet loss, jitter, Delay, and one way call. Alert is triggered once one of the threshold is reached and number of incidents is greater than the value or number of CDR is over percent threshold. Here is alert example which is triggered if 2% of calls is below MOS 3.5.



SIP response alert type triggers alerts based on SIP response type. In this example alert is sent if number of all calls with 5XX SIP response exceeds 2%



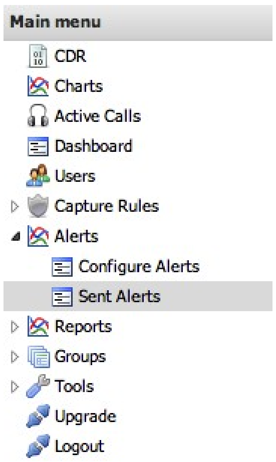


IP/Number group – choose to which group of IP/Numbers the alert is applied. Groups are defined in Groups main menu.

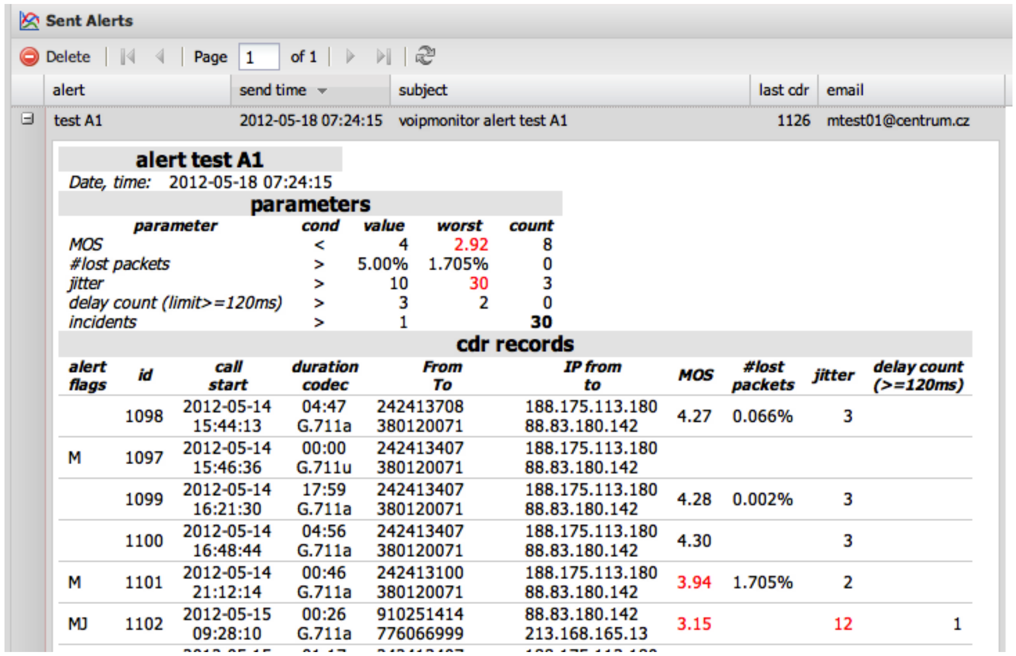
IP address/Numbers – choose individual IP addresses/numbers or network ranges to which is the alert applied. Delimited by [enter]

E-mail Group – choose to which Emails defined in groups should be alert sent.

E-mails – choose individual list of E-mails for alert delivery. Delimited by [enter].

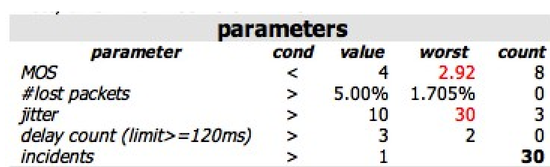


Sent alerts



Each sent alert is saved into history and looks exactly same as delivered in email.

In parameters table overall QoS metrics are shown with highlighted bad values.

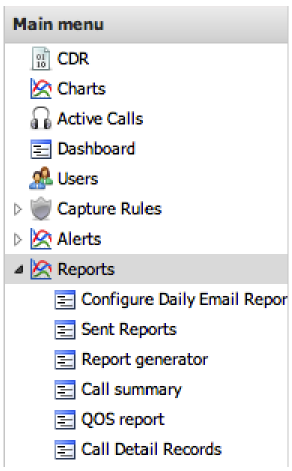


CDR records table shows individual cases. Alert flag column shows if the call alerted because of (M)OS, (J)itter, (L)oss or (D)elay.

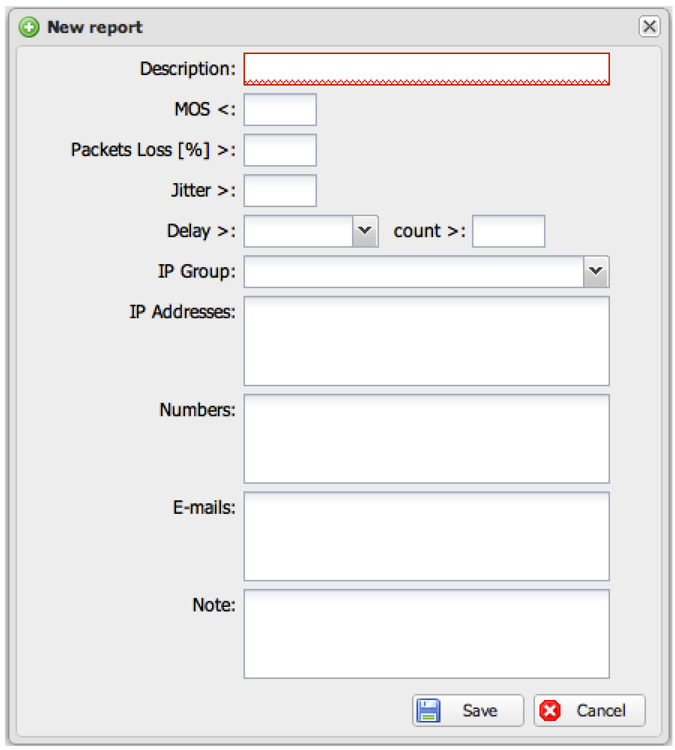
Reports

Reports contains daily reports, instant report generator, Call summary, QoS report and CDR simplified view.

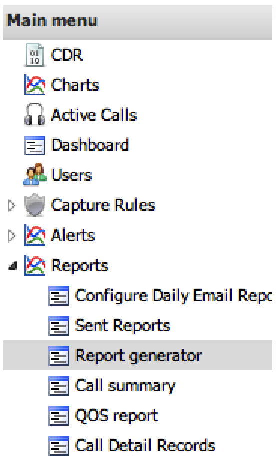
Daily Email Reports



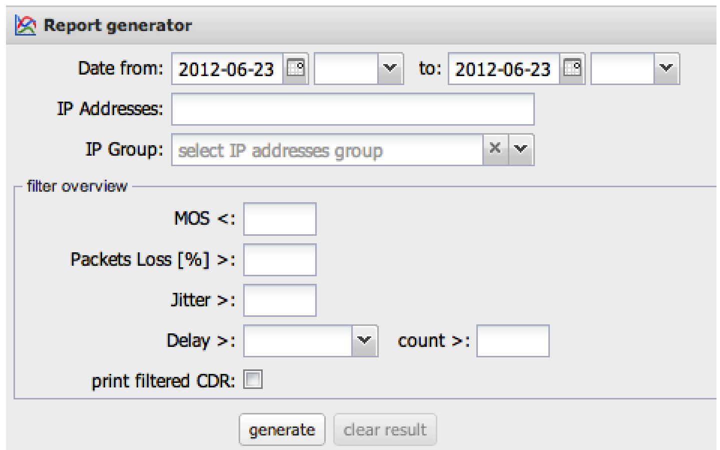
Daily email Reports is the same as in alerts in previous chapter with difference the report is sent once per day.



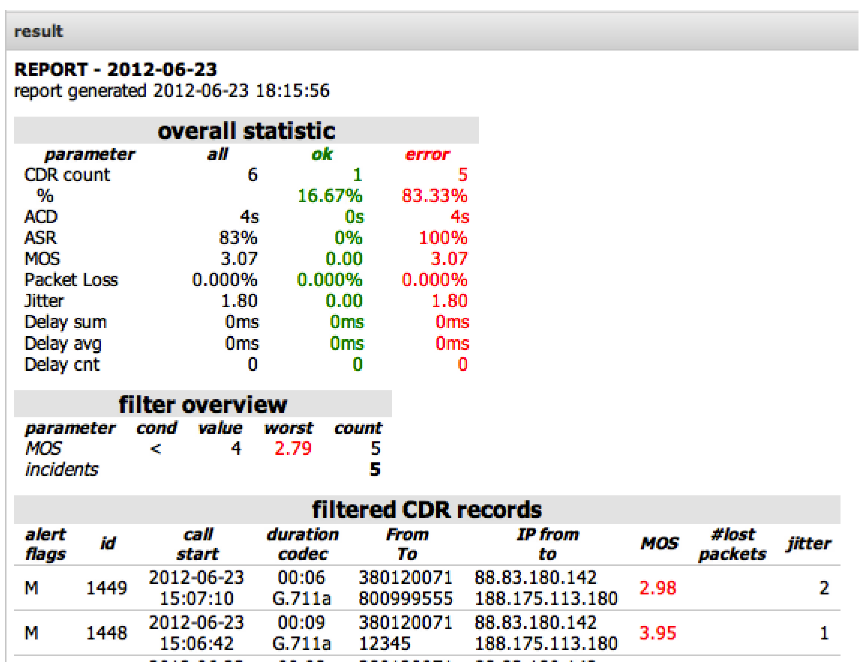
Report generator



Report generator allows create report from historical data based on various criteria.



After choosing Date, IP ranges and QoS parameters table with results shows up below the form.


result

REPORT - 2012-06-23
report generated 2012-06-23 18:15:56

overall statistic

parameter	all	ok	error
CDR count	6	1	5
%		16.67%	83.33%
ACD	4s	0s	4s
ASR	83%	0%	100%
MOS	3.07	0.00	3.07
Packet Loss	0.000%	0.000%	0.000%
Jitter	1.80	0.00	1.80
Delay sum	0ms	0ms	0ms
Delay avg	0ms	0ms	0ms
Delay cnt	0	0	0

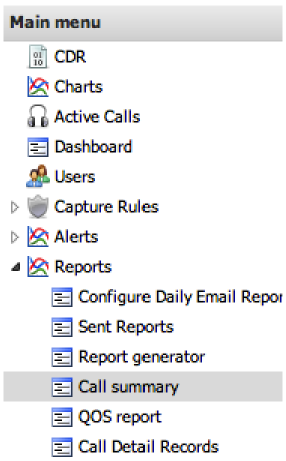
filter overview

parameter	cond	value	worst	count
MOS	<	4	2.79	5
incidents				**5**

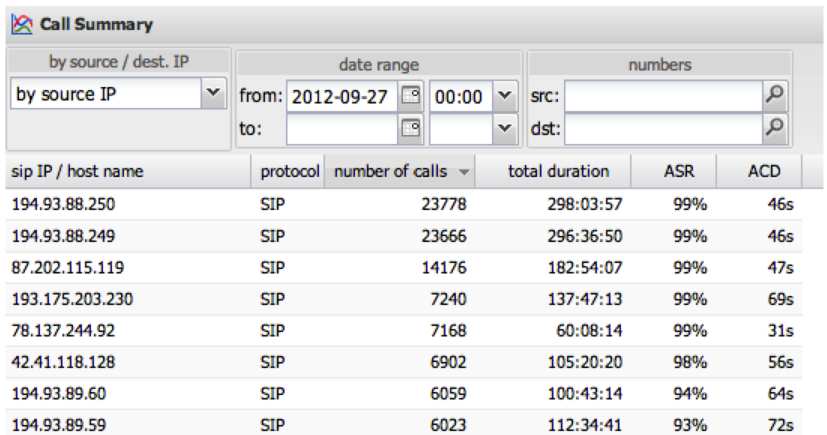
filtered CDR records

alert flags	id	call start	duration codec	From To	IP from to	MOS	#lost packets	jitter
M	1449	2012-06-23 15:07:10	00:06 G.711a	380120071 800999555	88.83.180.142 188.175.113.180	2.98		2
M	1448	2012-06-23 15:06:42	00:09 G.711a	380120071 12345	88.83.180.142 188.175.113.180	3.95		1

Call summary

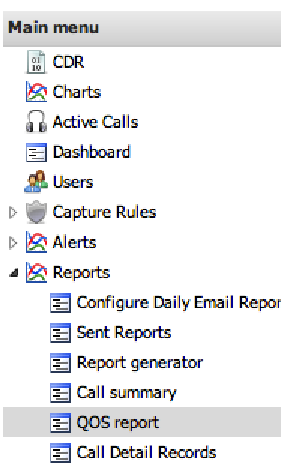


Call summary is brief overview grouped by IP source/destination IP addresses focused on signalling quality metrics including ASR, ACD, Total duration and total number of calls. Toolbar can be used to search by date range and also filter calls by source or destination numbers.

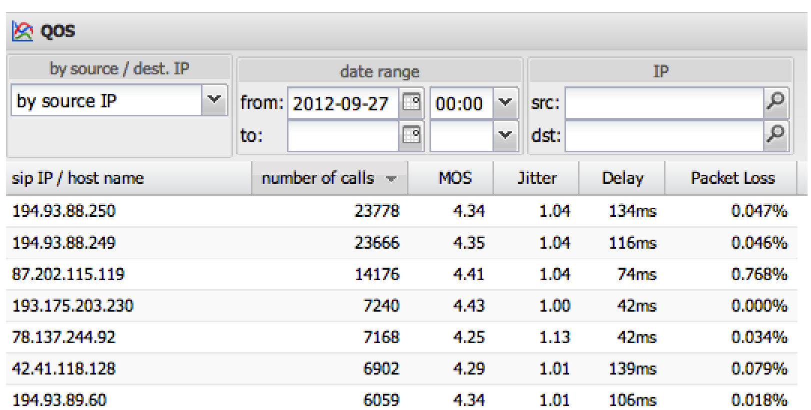


sip IP / host name	protocol	number of calls	total duration	ASR	ACD
194.93.88.250	SIP	23778	298:03:57	99%	46s
194.93.88.249	SIP	23666	296:36:50	99%	46s
87.202.115.119	SIP	14176	182:54:07	99%	47s
193.175.203.230	SIP	7240	137:47:13	99%	69s
78.137.244.92	SIP	7168	60:08:14	99%	31s
42.41.118.128	SIP	6902	105:20:20	98%	56s
194.93.89.60	SIP	6059	100:43:14	94%	64s
194.93.89.59	SIP	6023	112:34:41	93%	72s

QoS report

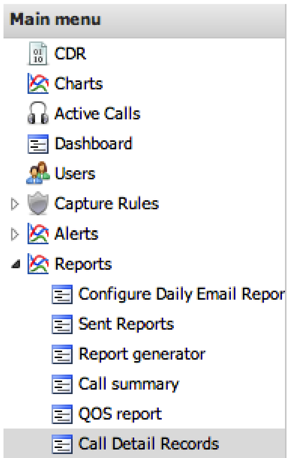


QoS report is simmiliar to Call summary but focused more on RTP statistics like MOS, Jitter, Delay and Packet loss. Toolbar can be used to filter by date range and IP range.

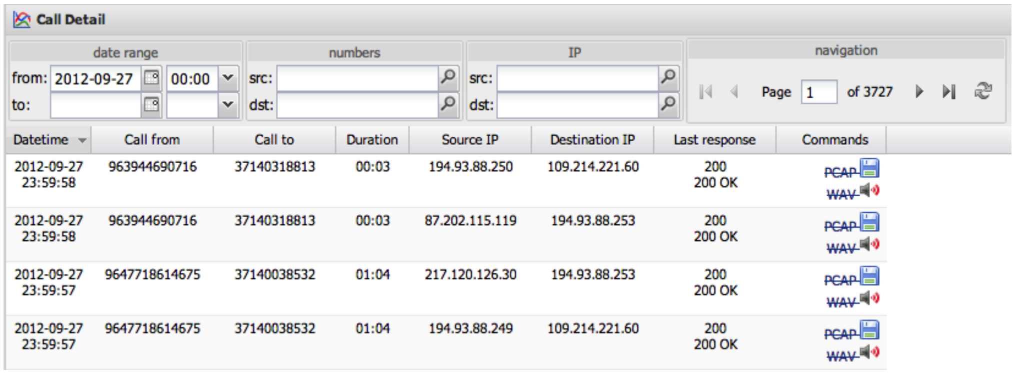


sip IP / host name	number of calls	MOS	Jitter	Delay	Packet Loss
194.93.88.250	23778	4.34	1.04	134ms	0.047%
194.93.88.249	23666	4.35	1.04	116ms	0.046%
87.202.115.119	14176	4.41	1.04	74ms	0.768%
193.175.203.230	7240	4.43	1.00	42ms	0.000%
78.137.244.92	7168	4.25	1.13	42ms	0.034%
42.41.118.128	6902	4.29	1.01	139ms	0.079%
194.93.89.60	6059	4.34	1.01	106ms	0.018%

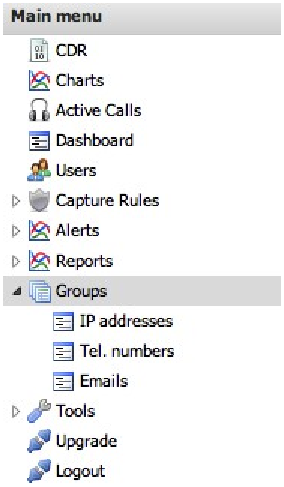
Call detail Records



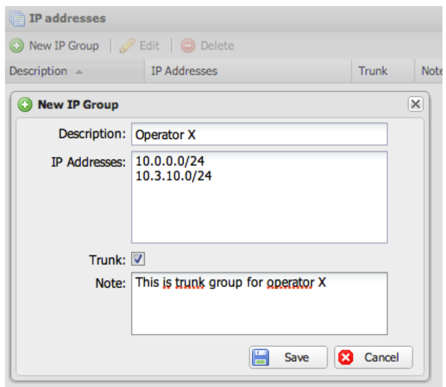
Call detail records is simplified interface to CDR showing IP and numbers with quick toolbar filters.

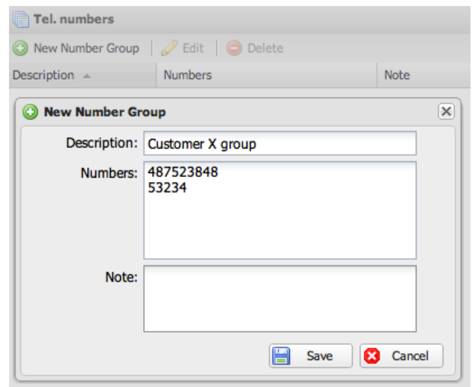


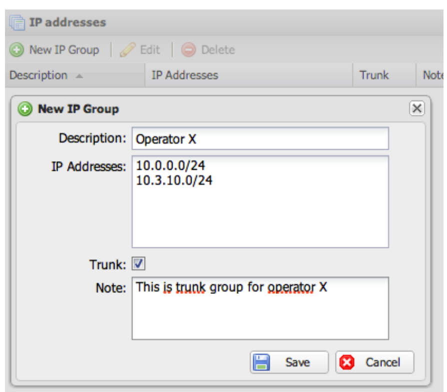
Groups



Groups defines set of IP addresses/networks, set of Tel.numbers / prefixes and set of Emails. Those groups can be used in several places across the entire WEB GUI. Typical is to define all SIP trunks from some operator as a group which can be used in Alerts or Filters. IP groups allows to check Trunk checkbox which is used to distinguish between internal/incoming/outgoing calls in CDR filters.



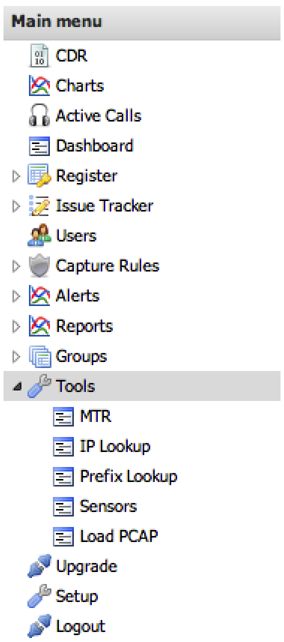


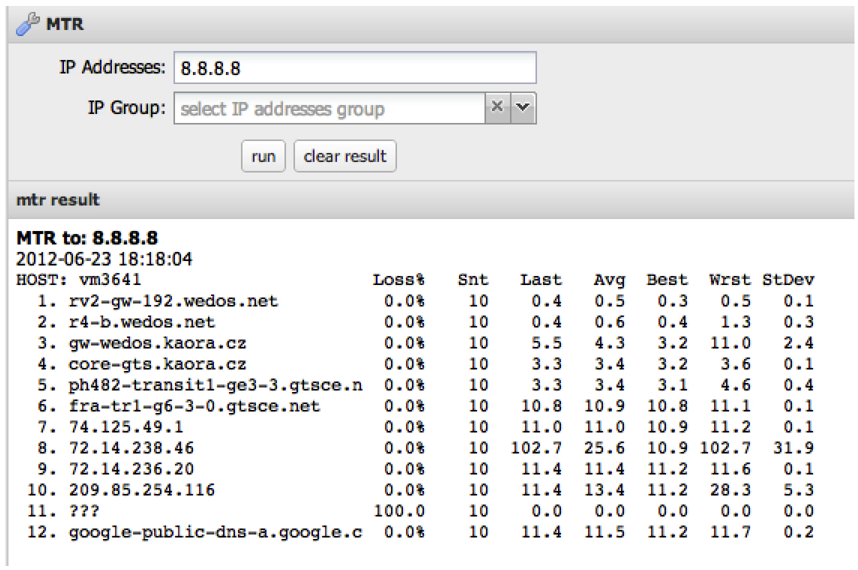


Tools

MTR shows trace from VoIPmonitor WEB server to selected IP address. The output is from linux mtr tool whis runs for 10 seconds and sends 10 packets.

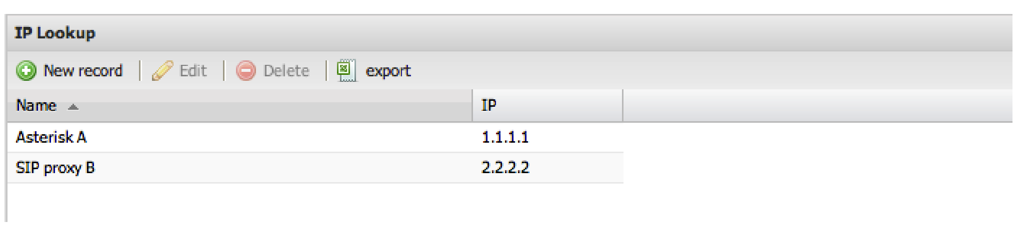
MTR





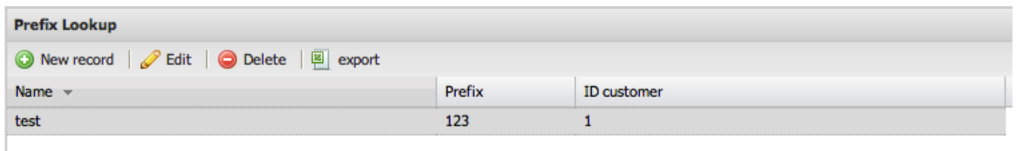
IP lookup

IP lookup table is used to substitute IP addresses in various places like CDR view. IP lookup table takes precedence over the DNS. To enable IP lookup you have to set ENABLE_SQL_IP_REVERSE_LOOKUP to true in config/configuration.php



Prefix lookup

Prefix lookup table is used to substitute numbers in various places like CDR view. Prefix lookup table takes precedence over the IP lookup and DNS. To enable IP lookup you have to set ENABLE_SQL_CUSTOMER_PREFIX_LOOKUP to true in config/configuration.php



Sensors

If you want to be able to read data from remote sensors or to be able to use "Legs by [CID|header]" in CDR detail - define here all sensors.

Sensor ID is number defined in /etc/voipmonitor.conf id_sensor = N
Name: is name of the sensor
Manager IP, Port is used for fetching data like pcap / graph files
Remote database parameter is used for trying to find relevant legs in "Legs by [CID|header]" CDR detail tab. This is usefull if you sniff the same legs / calls on various place of your network and you want to see all legs for a CDR.

Load pcap

Here you can upload pcap file captured by any tool using libpcap format which is tcpdump tshark wireshark voipmonitor and much more. Uploaded pcap file is read by voipmonitor:

voipmonitor --config-file /etc/voipmonitor.conf -r upload.pcap

where /etc/voipmonitor.conf can be changed in config/configuration.php constant UPLOADPCAP_SNIFFERCONF

Upgrade

Upgrade from version 5.X

The upgrade process is fully automatic and no user action is needed.

Upgrade from version 4 to 5

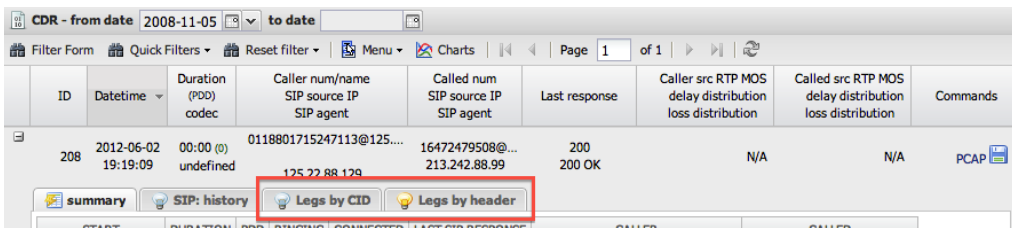
VoIPmonitor GUI version 5 has new database structure and is compatible only with sniffer version 5. Upgrading database is described in sniffer manual.

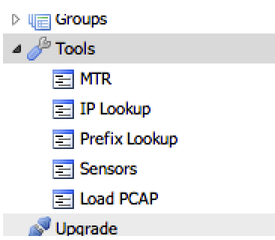
Configuration file – when upgrading from previous versions 4.X the new config/configuration.php has to be copied from config/configuration-template.php which is done automatically when doing new installation through web browser.

Whats new

5.2 --> 5.3 (build 429)

New Legs by CDR and Legs by header tabs in CDR detail. See CDR section.





New tools – prefix / IP lookup, sensors definition and load pcap. Check Tools section.

New API – http://server/api.php