Release Notes

Stacked Brains

VoIP Analyzer Tool

for macOS

https://www.voipanalyzertool.com/

Online Wireshark Analysis :

https://online.voipanalyzertool.com/

Release Notes Version: v1.19

Software Release is identified by

Version: v26.01.16.00

Notice:

This document contains confidential information that is proprietary to Stacked Brains.

No part of its contents may be used, copied, disclosed, or conveyed to any party in any manner whatsoever without prior consent.

DECLARATION DATE: January 16, 2026

Stacked Brains, Peiskeshof 76, 8790 Waregem, Belgium

https://www.stackedbrains.com/

info@stackedbrains.com

The information provided in this document contains merely general descriptions or characteristics of performance which in case of actual use do not always apply as described or which may change as a result of further development of the products. An obligation to provide the respective characteristics shall only exist if expressly agreed in the terms of contract. Availability and technical specifications are subject to change without notice. All other company, brand, product and service names are trademarks or registered trademarks of their respective holders.

1 Contents

1 History of change 5

1.1 History of change (Release Notes) 5

1.2 Product version history 6

2 Changes 7

2.1 New functionality 7

2.1.1 v26.01.16.00 7

2.1.2 v25.12.06.00 8

2.1.3 v24.06.01.00 10

2.1.4 v24.05.30.00 10

2.1.5 v24.05.10.00 10

2.1.6 v23.11.20.00 11

2.1.7 v23.11.02.00 11

2.1.8 v23.10.20.00 12

2.1.9 v23.09.16.00 12

2.1.10 v23.06.10.00 13

2.1.11 v23.04.21.00 13

2.1.12 v22.12.02.00 13

2.1.13 v22.06.01.00 15

2.1.14 v22.05.16.00 16

2.1.15 v22.03.14.00 16

2.1.16 v22.02.14.00 18

2.1.17 v21.12.27.00 20

2.1.18 v21.10.02.00 21
2.1.19 v21.09.16.00 21
2.1.20 v21.03.31.00 21
2.1.21 v21.02.22.00 22
2.1.22 v20.11.14.00 22
2.1.23 v20.10.27.00 22
2.2 Implemented Change Requests (CR's) 23
2.2.1 CR's 23
3 Hardware and software compatibility 23
3.1 Supported Operating systems 23
4 Important Information 23
4.1 Installation 23
4.1.1 Package dependencies 24
4.1.2 Install image 26
4.1.2.1 Big Sur 11.0 28
4.2 Run executable 32
4.3 Uninstall / Repair 33
4.4 Upgrade / Update 33
4.4.1 Checking version 33
4.5 Fallback 34
4.6 Security Considerations 35
4.7 Special settings and instructions 35
4.7.1 Configuration Requirements / Restrictions 35

4.7.1.1 Licensing 35
4.7.1.2 XML configuration file 37
4.7.2 VoIP Analyzer Tool Performance and Resilience 40
4.7.2.1 Vulnerabilities 40
5 Reported Problems / Symptoms under Analysis 40
6 Restrictions, Workarounds and Hints 40
6.1 Restrictions 40
6.1.1 Restrictions or Exceptions 40
6.2 Workarounds / Hints 40
6.2.1 Workarounds 40
6.2.2 Hints 40
6.2.2.1 XML configuration 40
7 Service information 42
7.1 License management 42
7.1.1 Open Source Software 42
7.2 Warranty 42
7.3 Remote serviceability 42

1 History of change

1.1 History of change (Release Notes)

Release Notes Version	Date	Changes

1.00	31 March 2021	First release notes of VoIP Analyzer tool v21.03.31.00 for macOS (beta-version)
1.01	16 September 2021	Release of VoIP Analyzer tool v21.09.16.00
1.02	2 October 2021	Release of VoIP Analyzer tool v21.10.02.00
1.03	27 December 2021	Release of VoIP Analyzer tool v21.12.27.00
1.04	14 February 2022	Release of VoIP Analyzer tool v22.02.14.00
1.05	14 March 2022	Release of VoIP Analyzer tool v22.03.14.00
1.06	16 May 2022	Release of VoIP Analyzer tool v22.05.16.00
1.07	1 June 2022	Release of VoIP Analyzer tool v22.06.01.00
1.08	2 December 2022	Release of VoIP Analyzer tool v22.06.01.00
1.09	21 April 2023	Release of VoIP Analyzer tool v23.04.21.00
1.10	10 June 2023	Release of VoIP Analyzer tool v23.06.10.00
1.11	16 September 2023	Release of VoIP Analyzer tool v23.09.16.00
1.12	20 October 2023	Release of VoIP Analyzer tool v23.10.20.00
1.13	2 November 2023	Release of VoIP Analyzer tool v23.11.02.00
1.14	20 November 2023	Release of VoIP Analyzer tool v23.11.20.00
1.15	10 May 2024	Release of VoIP Analyzer tool v24.05.10.00
1.16	30 May 2024	Release of VoIP Analyzer tool v24.05.30.00
1.17	1 June 2024	Release of VoIP Analyzer tool v24.06.01.00
1.18	6 December 2025	Release of VoIP Analyzer tool v25.12.06.00
1.19	16 January 2026	Release of VoIP Analyzer tool v26.01.16.00

1.2 Product version history

Software version	**Date**	**Comment**
v21.03.31.00	31 March 2021	released (beta-version)
V21.09.16.00	16 September 2021	released
v21.10.02.00	2 October 2021	released
v21.12.27.00	27 December 2021	released
v22.02.14.00	14 February 2022	released

v22.03.14.00	14 March 2022	released
v22.05.16.00	16 May 2022	released
v22.06.01.00	1 June 2022	released
v22.12.02.00	2 December 2022	released
v23.04.21.00	21 April 2023	released
v23.06.10.00	10 June 2023	released
v23.09.16.00	16 September 2023	released
v23.10.20.00	20 October 2023	released
v23.11.02.00	2 November 2023	released
v23.11.20.00	20 November 2023	released
v24.05.10.00	10 May 2024	released
v24.05.30.00	30 May 2024	released
v24.06.01.00	1 June 2024	released
v25.12.06.00	6 December 2025	released
v26.01.16.00	16 January 2026	released

2 Changes

2.1 New functionality

2.1.1 v26.01.16.00

- Support AudioCodes Call CDR records
- Support AudioCodes Media CDR records
- Support AudioCodes SDR records
- Adapt Audit handling after midnight
- Syslog listening port for service in registry
- siplog2pcap: support OpenScape Desktop Client Personal Edition sip logs
- siplog2pcap: correct OpenScape Desktop Client Personal Edition time format
- Service: Show summary of detected SIP interfaces at start
- Service: empty queue when service is stopped
- Service: optimize thread handling

- SQL: add database version schema
- SQL: optimize Day prefix handling
- SQL: rework of SIP useragent/diversion SQL query
- SQL: restart connection on exception
- SQL: correction on day independable tables
- SQL: Closing and opening SQL database only via the SQL thread
- SQL: improve GUI handling
- Support Graphs and Reports as output
- Show thread ID

2.1.2 v25.12.06.00

- Bug fixes and corrections

- siplog2pcap : new formats + corrections
- siplog2pcap: support Ribbon SBC sip logs
- siplog2pcap: support live Unify IP Phone syslog to pcap
- siplog2pcap: support Oracle Acme Packet SBC sip logs
- siplog2pcap: support Polycom SIP logs
- siplog2pcap: support Blink SIP logs
- siplog2pcap: support OpenScape mediaserver
- siplog2pcap: vendor Grandstream

- text2pcap : new formats + corrections
- text2pcap: three digit prefix format

- syslog2pcap: support fragmented SIP messages
- syslog2pcap: support CDRs AudioCodes
- syslog2pcap: write readme.txt for Wireshark pipe

- Service for syslog capturing
- Service control via GUI

- Processing service handling at midnight

- SQL database introduction (MySQL/MariaDB)
- Day selection for database
- Filtering for audit from SQL database
- Audit to SQL database (MySQL/MariaDB)

- Improved xml config handling (also save xml)
- SIP error codes configurable in xml
- Word/pdf output configurable via XML

- Separating Audit processing and audit reporting (output)
- Handling of incomplete SIP calls
- New code signing certificate (Certum)
- Align gen_Logging code between GUI/CMD/SRV
- Change internal certificate for license
- Separate threads for syslog to pcap/pipe/sql
- Improvements on corrupted packet handling
- Flush of pcapng file
- Python code for making the graphs + example markdown file for generating the report
- Word/pdf output in separate file
- Password encrypting/decrypting
- Rework of CSTA message handling: dictionary and splitting of contents
- Rework of exception handling
- Rework of DLL handling in building the software
- Richtextbox improvements for main Form
- Detect SIP BYE exact 20 seconds after SDP SendOnly attribute was sent
- Anonymizing for GSM/ISDN bit-oriented protocols
- String concatination performance improvement
- Reorder of pcap/pcapng files before audit

2.1.3 v24.06.01.00

- Make build for Windows + Linux + macOS

2.1.4 v24.05.30.00

- Anonymize additional parameters for following protocols: mysql, stun
- Mark SIP calls with a missing SIP ACK as an incomplete call
- Optimize extracting of UDP and TCP

text2pcap: autodetect all possible

2.1.5 v24.05.10.00

- text2pcap: support lots of text files with new hex formats
- text2pcap: autodetect all possible pcap link layers
- text2pcap: write pcapng format with possible different linklayers
- text2pcap: support the OSIX format
- text2pcap: support vlan tagged frames
- text2pcap: optimize processing of Juniper hex dump logs
- text2pcap: optimize processing of LTE hex dump logs
- text2pcap: optimize rawIP detection
- text2pcap: optimize detection of 0000 prefixes in hex dump
- flexible anonymize: anonymize MAC addresses to "private" OUI
- flexible anonymize: anonymize all IMSI fields (any layer, any protocol)
- flexible anonymize: anonymize LLDP
- flexible anonymize: anonymize the SCTP protocol
- flexible anonymize: anonymize IPv6 in IPv4 (6to4)
- flexible anonymize: anonymize IPv4 in IPv6: support Teredo tunneling
- flexible anonymize: remapmac for ssl.src.eth
- flexible anonymize: correct VRRP checksum
- flexible anonymize: correct FPP checksum
- flexible anonymize: leave 127.0.0.0/8 original during anonymize
- siplog2pcap: support TEL URI
- siplog2pcap: optimize Grandstream SIP logs

- siplog2pcap: support detection of SIP INFO messages
- merge: write pcapng format after merge
- reoder: support the reoder of pcap files in GUI
- code signing: codesigning of executable with Certum certificate

2.1.6 v23.11.20.00

- Remap all Wireshark fields with IPv4 field type
- Remap all Wireshark fields with Ethernet field type
- Remap IPv4 addresses for the OSPF protocol
- Optimize mapping of MSISDN to anonymous phone numbers
- Optimize ICMP checksum calculation during anonymize
- Optimize calculation of CRC frame checksums after anonymize of a pcap file
- Optimize TCP segmentation during pcap anonymisation
- Optimize broadcast and multicast MAC addresses during anonymisation
- Optimize anonymisation of SIP TEL URI
- Support anonymisation of gsm-r-uus1.elda: GSM: Enhanced Location Dependant Addressing
- Anonymize "sip.r-uri" also when there is no .user or .host part
- Support anonymisation of the sdp.phone parameter
- Show statistics at the end which fields were anonymized
- Siplog2pcap: Unify OpenScape Business: detect source and destination IP address for SIP messages

2.1.7 v23.11.02.00

Added “flexible anonymize”: anonymize ANY field at ANY protocol at ANY level. No limits.

Just add the name of the “Wireshark display filter” of the field to be anonymize in our anonymize_display_filters.csv file. The file is already pre-filled with almost 300 Wireshark display filters, but can easily be extended with additional Wireshark display filters which fields needs to be anonymized.

- remap all possible MAC addresses to anonymous MAC addresses (on any layer from)

- remap all possible IPv4 fields to anonymous IPv4 fields (on any layer)

- anonymize all possible hostnames conform presentation of Betty Dubois on SharkFest23EU in Brussels

- correct flexible anonymize processing after feedback on SharkFest23EU in Brussels

2.1.8 v23.10.20.00

Added some very unique features. Quite unique in the market:

Support configurable CSV text file so that ANY field, ANY parameter, ANY protocol can be anonymized/obfuscated for ANY Wireshark PCAP file.

You only have to add the Wireshark display filter name in the new CSV text file.

Several parameters inside following protocols are already by default anonymized:

Supported protocols in alphabetical order:

- a11, arp, browser, capwap, cdp, cflow, dhcp, dhcpv6, diameter, dns,, dtp, e164, eap, eth, ftp, gsm, gtpv2, hsrp, hsrp2, http, icmpv6, imf, inap, ip, ipcp, ipx, isup, Kerberos, lacp, ldap, ldp, lldp, mgcp, nbdgm, nbns, netbios, ntp, ospf, pppoed, radius, rip, rtcp, rtp, sccp, sdp, sip, sll, smb,smb2, smtp, stp, syslog, telnet, tftp, tls, udid, uma, vrrp, vtp, whois, wlan
- Additional protocols or fields can be added easily to be anonymized

- Support re-mapping of any IPv4 IP address in ANY Wireshark field of type IPv4.
  Re-mapping is done to an anonymized IP Address. Same IP address is consequent re-mapped to the same anonymized IPv4 address.
  Currently following protocols trigger an IP address remapping: arp, dhcp, dns, ip, sctp, uma vrrp. Other protocols can be added easily.
- Support re-mapping to anonymous values of ANY MSISDN phone number in ANY parameter in ANY protocol
  (also 3G/4G/5G protocols, as also in following protocols: e164, gsm, isup, sccp)
  Same MSISDN phone number is consequent re-mapped to the same anonymized MSISDN phone number
- Support re-mapping to anonymous values of ANY IMSI number in ANY IMSI parameter in ANY protocol
  Same IMSI number is consequent re-mapped to the same anonymized IMSI number
- Support re-mapping to anonymous values of ANY IMEI number in ANY IMEI parameter in ANY protocol
  Same IMEI number is consequent re-mapped to the same anonymized IMEI number

2.1.9 v23.09.16.00

- **Optimize SharkEdit:** Anonymize any field in any Wireshark pcap file based on the well known Wireshark display filters
  - also called “flexible anonymize” in the GUI

- **Optimize SharkExport:** Export/filter any field in any Wireshark pcap file based on the well known Wireshark display filters. Export on HTML format and in Excel CSV format
    - also called “flexible filter” in the GUI
- Support LinkLayers.Ppp
- text2pcap: add syslog remark when limit of 1000 packets is reached
- text2pcap: better detection of date/time for unknown vendors
- Bug fixing

2.1.10 v23.06.10.00

- **NEW: SharkEdit:** Anonymize any field in any Wireshark pcap file based on the well known Wireshark display filters
    - also called “flexible anonymize” in the GUI
- **NEW SharkExport:** Export/filter any field in any Wireshark pcap file based on the well known Wireshark display filters. Export on HTML format and in Excel CSV format
    - also called “flexible filter” in the GUI
- Add live suggestion in GUI of any Wireshark display filter for anonymize or export/filter
- Bug fixing

2.1.11 v23.04.21.00

- bug fixing

2.1.12 v22.12.02.00

- **SIP**
    - Siplog2Pcap: conversion from a SIP log to a pcap now supported for following SIP vendors:
        - 3CX
        - Alcatel
        - Anveo
        - Asterisk
        - AudioCodes Syslog
        - AudioCodes SIP ladder diagram files
        - Avaya

        - BroadWorks XS logs (BroadSoft)
        - Cisco or CUCM Sip Logs
        - Enovation UMO (UMO SIP Alarm devices)
        - Genesys
        - Lync
        - Mediatrix Syslog
        - Microsoft Teams -> see AudioCodes syslog
        - Mitel
        - OneAccess (Ekinops)
        - OpenScape Business (Atos/Unify)
        - OpenScape 4000 STMI SIP logs (Atos/Unify)
        - OpenScape Voice RTT traces (Atos/Unify)
        - RingCentral
        - Twilio
        - Yealink
    - Other SIP vendors can be supported on simple demand.

    - SipLog2Pcap: OpenScape voice RTT trace: optimize handling of SIP-Q message body
    - Show also SIP CANCEL after SIP RE-INVITE as incomplete SIP dialog
    - Show SIP Source IP and SIP Destination IP as summary in SIP Ladder diagram pop-ups
    - SIP Call duration is zero when there is no SIP BYE message

- **Anonymize**
    - anonymize the HTTP protocol
    - anonymize the SSDP protocol
    - anonymize the FTP protocol (USER and PASS only)
    - anonymize also SIPS (Secure SIP)
    - Text2Pcap
    - Text2Pcap: make detection of start of a new hex packet more defensive
    - Text2Pcap: support Cisco ASR 5000 / Cisco ASR 5500 hexdump file format (RawIP)
    - Text2Pcap: support Fortinet hex dumps

    - Text2Pcap: support "Cisco monitor capture" format
    - Text2Pcap: support Juniper Netscreen Snoop format hex dumps

- **Other**
    - Support CSTA III during pcap analysis: Show CSTA III messages in generated HTML page
    - optimize readability of the XML protocol in a SIP message body in the SIP ladder diagrams
    - support IP fragmentation for IP in a GRE tunnel (Generic Routing Encapsulation)
    - Optimize SEO
    - support TZSP encapsulation: TaZmen Sniffer Protocol
    - Generate wave files also for SRTP
    - Optimize RTCP logging
    - support of 802.3br packets

2.1.13 v22.06.01.00

- SIP
    - optimize creation of short SIP headers when there is a ";" sign in the SIP header display part
- RTP
    - RTP comfort noice not correct indicated for SRTP in RTP HTML table
- Anonymize
    - anonymize FTP password
    - anonymize HTTP GET string
    - always anonymize SIP TEL URI (also when they are not RFC conform formatted)
    - keep original IP addreses during anonymize when there are pre-defined documentation IP addresses (RFC5737)
- Automatic update
    - check version.txt and version.html at startup
    - detect download of update failure. Make timeout shorter.
    - optimize handling if HTTPS request to check latest version is blocked
- GPRS Tunneling Protocol (GTP)
    - optimize detection and unpacking of of GPRS Tunneling Protocol (GTP) when there are optional GTP parameters
- other

    - show license request sting in the logging at startup when there is no license
    - adjust executable names for linux (all lower case now)
    - default path to c:\voipanalyzer in Windows install program

2.1.14 v22.05.16.00

- SIP
    - anonymize RFC7315 private SIP headers
    - support RFC3262 RAck SIP header for ABNF check
    - SIP request line sometimes too long in the SVG diagram
- AudioCodes
    - Major performance optimalisation for AudioCodes SipLog2Pcap conversion
    - optimize AudioCodes SID filtered files
    - Detect CallID2 for Audiocodes sipLog2PCap. Show in HTML tables CallID2
    - Support multipart body for SipLog2Pcap (e.g. QISG, SIP-Q, ...) (e.g. for OpenScape Voice - OpenScape 4000 - OpenScape Business)
- - RTP
    - anonymize RTP payload while volume is still detectable
    - detect ambiguous RTP sender timestamp
- - MGCP
    - defensive reading of MGCP messages. allow
 and \n
- - updater
    - add automatic online updater
- - ZIP
    - minor correction for unzipping with gzip
- - General
    - optimize TCP reassemble when reassembled packet is larger than 64kB
    - Add counter in HTML tables simular as in CSV files

2.1.15 v22.03.14.00

- SipLog2Pcap

    - add CONVERSIONS button in GUI
    - SipLog2Pcap: convert SIP logs to Wireshark pcap files. SIP log conversions supported from following vendors:
        - Anveo
        - Asterisk
        - AudioCodes Syslog
        - AudioCodes SIP ladder diagram files
        - Broadsoft
        - Genesys
        - Lync
        - Mediatrix Syslog
        - Mitel
        - OneAccess (Ekinops)
        - OpenScape Business (Atos/Unify)
        - OpenScape 4000 STMI SIP logs (Atos/Unify)
        - OpenScape Voice RTT traces (Atos/Unify)
        - Twilio
- AudioCodes
    - convert AudioCodes syslog files to an audit with a filtered AudioCodes syslog file for each unique SIP call [SID=xxx:xxx:xxx]
    - Add unique AudioCodes Log as table entry in all the HTML pages after audit
    - Support Audit of AudioCodes Debug Recording (ACDR) capture inclusive AudioCodes logs (SIP + RTP + RTCP)
    - Convert AudioCodes debug recording capture to a normal Wireshark capture during the Merge functionality (SIP + RTP + RTCP)
- Hex2Pcap: convert Wireshark HEX files to Wireshark pcap files
- Tunneled QSIG in SIP message body
    - more exact processing of tunneled QSIG in SIP (or SIP-Q): process as HEX
    - detect false hits on \r in multipart/mixed body
    - Support "Content-Disposition:" as body header for multipart/mixed
- Still show the tool GUI when the license is expired
- Optimize handling of captures on the Null/loopback device
- Use "frame.number==" in loggings (not WP== anymore)
- Correct gzip issue during merge on Linux
- Anonymize

    - also anonymize optional parameters in the SIP FROM header
    - support anonymize of IGMP multicast packets
    - do not process DHCP packets during anonymize
- RTP
    - Show Max RTP delay, max RTP Jitter and Max RTP Skew in RTP HTML table
    - correct influence between RTP and DTMF on RTP statistics
-

2.1.16 v22.02.14.00

- correct influence between RTP and DTMF on RTP statistics
- detect IPv6 For Raw captures
- add extra hints in syslog for anonymize
- anonymize ICE candidates in SDP
- SipLog2Pcap: paste DateTime also when year is not available (e.g. for AudioCodes syslog)
- support AVPF and SAVPF in SDP
- clean addition struct memory at the end of an audit/anonymize/merge
- make anonymized DNS names iso mapped IP addresses
- use more natural filename after anonymize (xxx_ANON)
- correct rewriting of SDP length after anonymize
- optimize TCP re-assembly mechanism
- optimize cleanup of leftovers after TCP assembly
- give warning when zip fle is opened
- correct processing of abbreviated headers
- TCP reassembly rework + hexdump in hex2pcap format
- gzip for pcapng
- cleanup/optimize IP fragmentation
- SipLog2Pcap generates incorrect "application/csta+xml" body
- optimize handling of pcapng capture with both ethernet packets as also sll packets
- Only last RTP codec is shown in the HTML RTP overview
- Update logging for abbreviated SIP headers
- Support the SIP NEGOTIATE method

- optimize logging of sip.Call-ID=="xxx"
- Update SipLog2Pcap help file
- SipLog2Pcap: support Mitel Sip log conversion to pcap
- SipLog2Pcap: add Mitel and OpenScape Voice RTT log conversion to pcap
- Correct PCAP with wrong TCP flags (e.g. push flag)
- solve progress bar > 100% issue
- Support detection of IEEE802.3br preemption
- Add proprietary X-RTP-Stat headers
- LinkLayerType as short
- update on IEEE 802.3br detection
- Add proprietar Siemens SIP headers
- Add proprietar ThigSbc SIP "X-" headers
- SipLog2Pcap: add vendor OneAccess for SIP log conversion to pcap
- SipLog2Pcap: also detect abbreviated content-type SIP header "c:"
- Read QSIG in body as bytes, not as chars
- SipLog2Pcap code optimalisation: make separate classes
- support ^M
- SipLog2Pcap: detect splitted SIP headers and combine them
- Add also "SIP Reason header" in HTML table for Register, Options etc
- Show SIP 403 Forbidden in the HTML ERROR tables
- SipLog2Pcap: detect SDP splitted on 2 lines for AudioCodes syslog logs.
- SipLog2Pcap: optimize splitted SIP messages
- SipLog2Pcap: support SIP messages on one line with numerous \n
- SipLog2Pcap: only search for \n if also SIP/2.0 can be found in the string
- add icons to Web projects and put header on index page
- SVG corrupted for "var protmessage0" when \r leftover occurs
- first detection SIP Common Log Format (CLF)
- error in cleanup tcp stream
- siplog2pcap html corrections (header)
- web: correct clear page for Firefox + make buttons
- Show errors/warnings when SIP RFC errors occur

2.1.17 v21.12.27.00

- Support "RAck" SIP header (RFC3262)
- support IP fragmentation with duplicate packets
- adjust handling of fragmented packets
- Adjust default OpenScape SBC ports from 50000-50019 to 50000-50039
- check maximum framesize of 64K after IP or TCP assembly
- do not use duplicate RTP packets for wave files
- cleanup at the end of IP fragments and TCP segments
- hex2pcap for online tool on website https://hex2pcap.voipanalyzertool.com/
- hex2pcap: detect de-chunked data and skip it
- allow a tab character for Hex2Pcap splitting
- allow maximal merged filesize of 4 Gb
- optimize link layer detection
- detect RTP packets when they use well known SIP ports
- allow any SIP response code in the range 100-699
- add remark in HTML when RTP delay is highter than 60ms
- check if the 5 mandatory SIP headers are available during audit
- prevent false hits on RTP and RTCP after SIP BYE
- SipLog2Pcap for online tool on website https://siplog2pcap.voipanalyzertool.com/
- online web tool: make specific page for wrong cases (e.g. wrong input)
- Wireshark OSPF packet not always correct parsed
- support radius and diagram protocol (not in GUI yet)
- adjust parsing of MGCP request line
- adjust parsing of CSeq SIP header
- do not allow SVG drawing with negative x position
- add help file for relevant commands for tshark to text
- culture dependency for DateTime
- use minimal HTML table height when table is empty
- show short names for wave files in HTML tables

- Add SIP reason header in generated HTML tables
- optimize splitting of long strings (e.g. SIP From/To/CallID) in HTML tables

2.1.18 v21.10.02.00

- now supported as Date format: day/month/year or month/day/year or year/month/day
- Support RawIP for Hex to Pcap conversions
- also anonymize SRTP
- give warning for zero duration parameter for DTMF via RFC2833/RFC4733
- use only destination IP and destination port for RTCP filenames
- check if SDP content length value header is consistent with SDP body length
- minor correction for new SDP content length after anonymize
- major performance optimalisation during writing of syslog entries from Pcap to syslog text files
- correction regarding detection of multiple SIP messages in one single Wireshark packet
- support P-RTP-Stat header for RTP statistics
- ignore X-Siemens-RTP-Stats: stats not available

2.1.19 v21.09.16.00

- major performance optimalisation during audit
- finetune anonymize of MGCP messages
- minor updates in help files
- create new CHM help file
- minor bug fixes

2.1.20 v21.03.31.00

- support of the MGCP protocol
- Handling SVG message graphs more generic (also for MGCP)
- Correction on script error popup for .chm help file
- Correction with multiple linktypes in pcapng
- Filter column buttons in HTML tables
- issue solving

2.1.21 v21.02.22.00

- support Raw IP header
- use colors in RTP HTML pages to highlight issues with packet drop, jitter, skew etc
- support abbreviated SIP headers
- use always TTL value 64 after anonymize to hide network topology
- support detection of GRE tunnel will NULL encryption
- add “clear logging window” button in GUI
- TCP segmentation finetuning
- Detect SIP BYE immediate after SIP RE-INVITE (incomplete SIP dialog)
- Make ethernet address anonymous during anonymize
- Add dropzone on website
- Added free online demo on our https://www.voipanalyzertool.com website
- Optimize performance
- support Ubuntu 20.10 (Ubuntu 20.04 was already supported)

2.1.22 v20.11.14.00

- performance optimalisation during merge action
- update of online help files
- update of offline CHM help files
- update tool icon
- bug fixes

2.1.23 v20.10.27.00

- Initial version released for any 64 bit Linux Operating system
- Supports same features as the Windows OS version
- Support of PCAPNG (pcap next generation) Wireshark format
- Support the Wireshark snoop format
- Add also a split button next to the merge button
- Improved performance in some area’s
- Add progress bar in GUI
- use UTC time for wireshark captures

- make also RTP stats in CSV format
- also anonymize data TCP packets with data length zero (e.g. TCP SYN)
- support merging of Ethernet packets + Linux SLL packets at the same time to one wireshark capture
- bug fixing

2.2 Implemented Change Requests (CR's)

2.2.1 CR's

None.

3 Hardware and software compatibility

3.1 Supported Operating systems

Currently following macOS distributions are supported :

- Big Sur 11.0

Is your macOS distribution or the exact version not listed here ?

Just send an email to macos@voipanalyzertool.com. We are glad to help you further.

4 Important Information

4.1 Installation

4.1.1 Package dependencies

After installation of the VoIPanalyzer Tool (see next sections) a help file is available in chm format (compressed html) and this can be viewed with a chm reader application. A chm reader application is not needed for the direct functionality of the VoIPAnalyzer Tool.

In the App Store you can search for chm :

chm

- chm viewer star
- chmod permissions tool
- chmpages
- chm-to-pdf
- chm viewer
- chm reader plus
- chm to pdf fast converter
- chmplus: chm reader
- chm view
- chm to pdf: the complete...

Develop

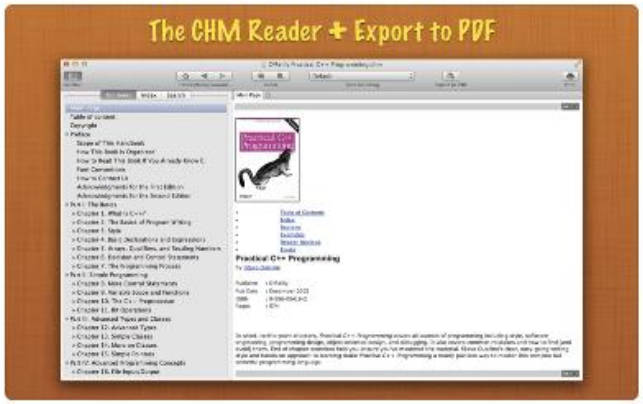
Categories

Updates

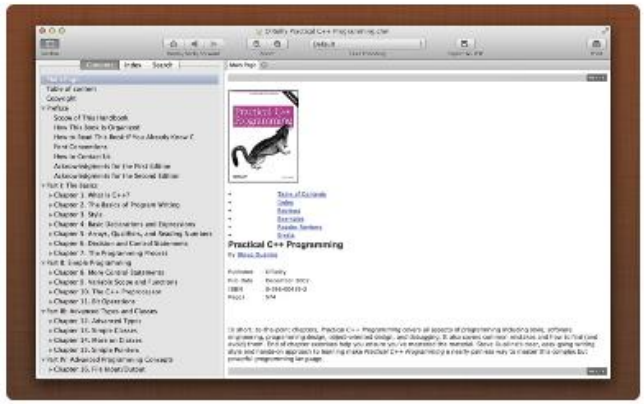
Results for “chm”

Filters

Read CHM+ : The CHM Read...
Utilities
3,49 €

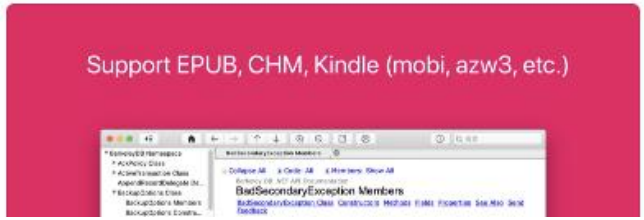


Read CHM
Utilities
★★★★☆ 1
OPEN



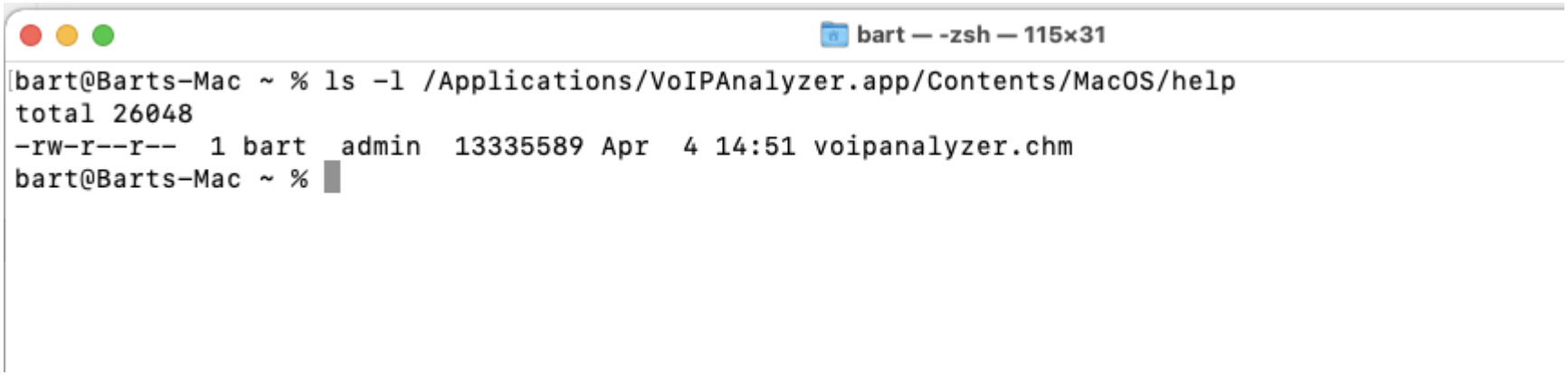
ChmPages
Productivity
GET

GMaster Reader
Read EPUB,CHM,Kindle, use it.
GET
In-App Purchases



After installation (see next section) the voipanalyzer.chm help file is under :

/Applications/VoIPAnalyzer.app/Contents/MacOS/help



4.1.2 Install image

The software can be downloaded from https://www.voipanalyzertool.com/download-macos

Download the install package. The downloaded install package is a .zip file that contains the macOS .dmg container for the VoIPAnalyzer Tool.

If necessary unzip the .zip file with :

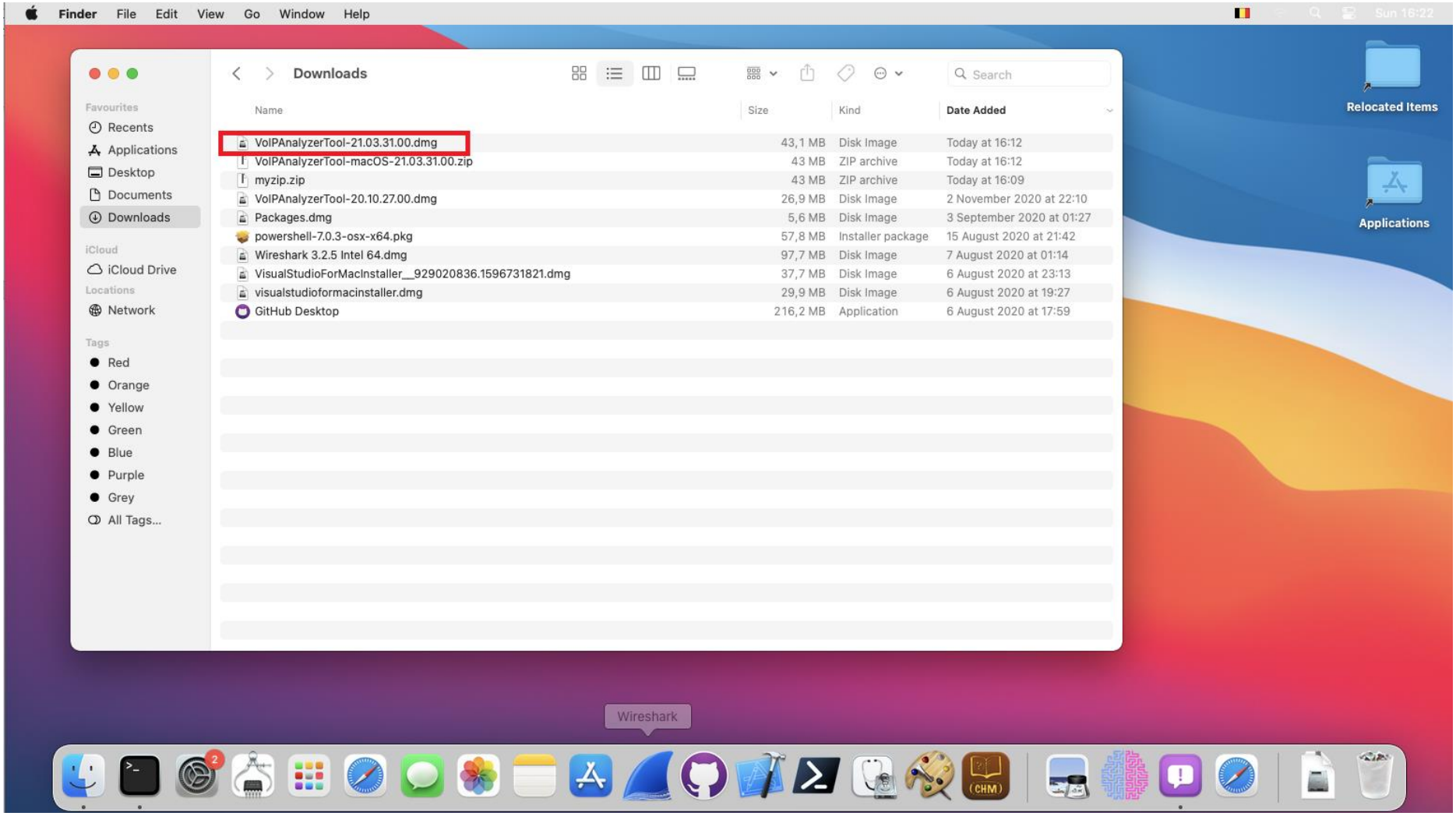
unzip *<VoIPAnalyzer Tool package name>*

(note: the package name can differ a bit from the screenshot, e.g. containing a different version number)

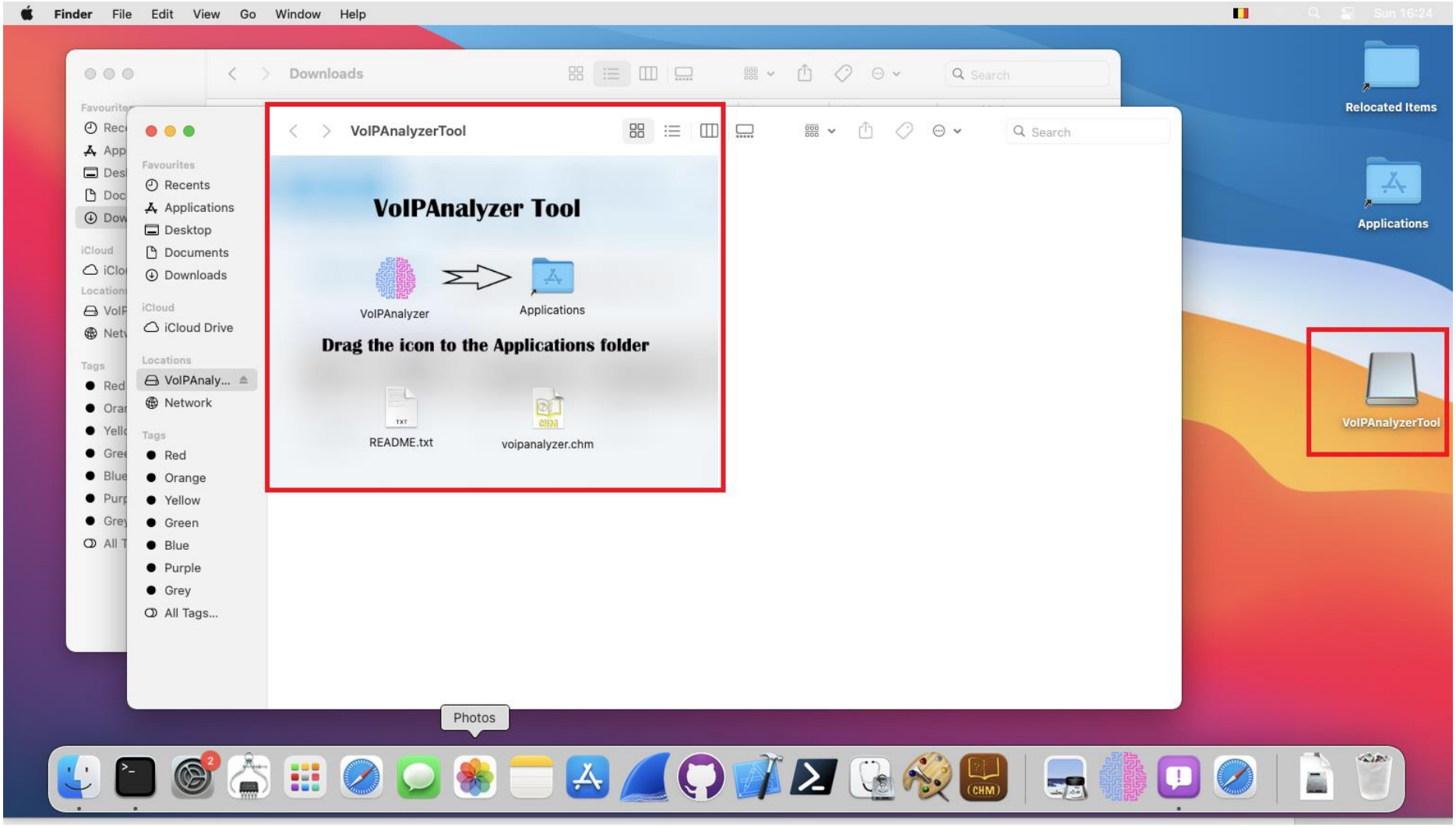
```
bart@Barts-Mac Downloads % unzip VoIPAnalyzerTool-macOS-21.03.31.00.zip
Archive:  VoIPAnalyzerTool-macOS-21.03.31.00.zip
replace VoIPAnalyzerTool-21.03.31.00.dmg? [y]es, [n]o, [A]ll, [N]one, [r]ename: y
  inflating: VoIPAnalyzerTool-21.03.31.00.dmg
bart@Barts-Mac Downloads %
```

4.1.2.1 Big Sur 11.0

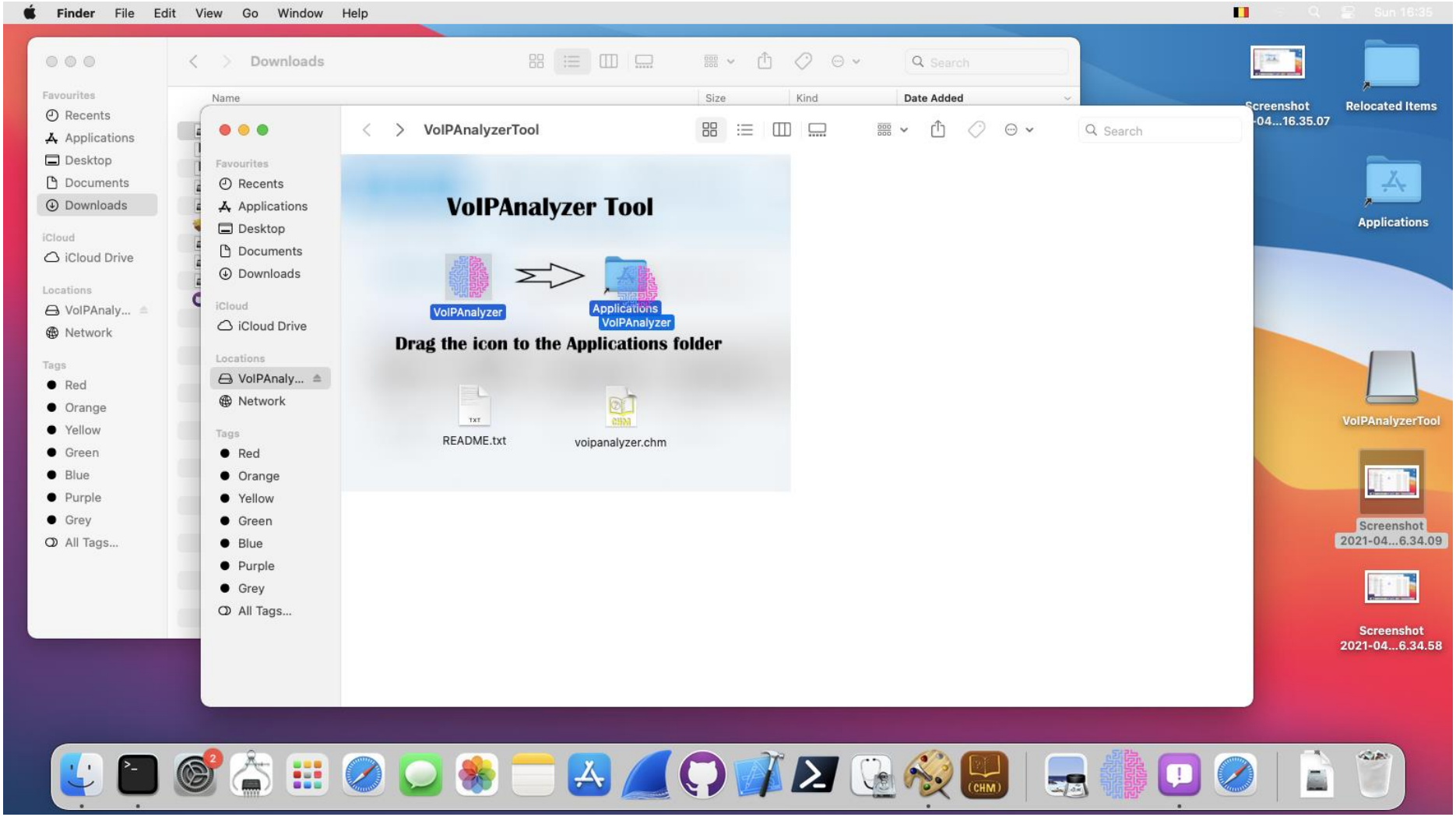
To install the .dmg file go with Finder to the download location :



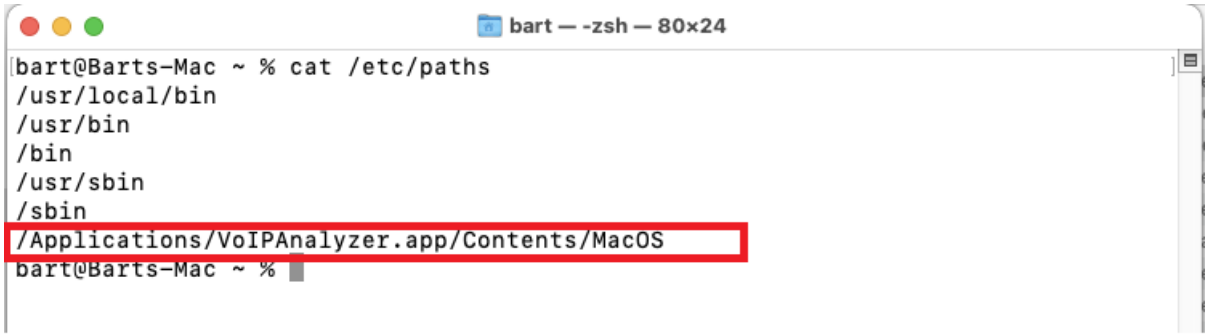
and double click the dmg file. This will open the container :



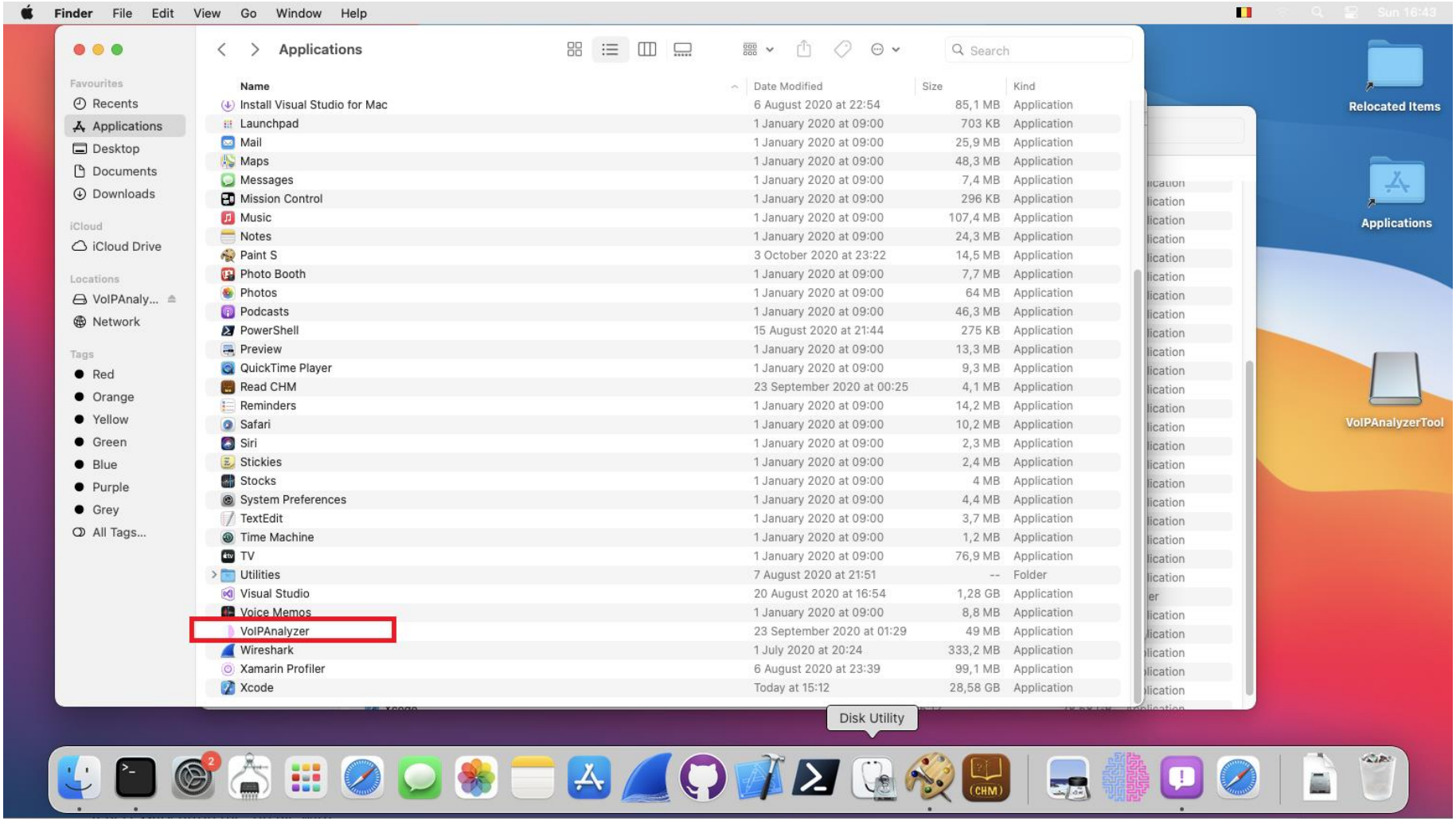
Just drag and drop the VoIPAnalyzer icon onto the Applications folder :



To run the VoIPAnalyzerCmd executable from everywhere add the correct path to the /etc/paths file :



The VoIPAnalyzer Tool will be shown under the Applications folder :



4.2 Run executable

To run the VoIP Analyzer Tool on the command line execute in a terminal window :

VoIPAnalyzerCmd

4.3 Uninstall / Repair

4.4 Upgrade / Update

Package upgrade / update is same as the installation and a popup will appear to replace the installed version.

4.4.1 Checking version

When starting the VoIP Analyzer Tool from the terminal window the version is shown :

```
bart@Barts-Mac Downloads % VoIPAnalyzerCmd

20210404_161854: ##########################################
20210404_161854: Program Version [21.03.31.00]
20210404_161854: ##########################################

Reading the XML Config file parameters at
[/Users/bart/.voipanalyzer/VoIPAnalyzer_xmlconfig.xml]

No filter active during merge/split based on Source or Destination IP addresses of Wireshark packets.
XMLConfig: process SIP     INVITE  = true (always)
XMLConfig: process SIP   REGISTER  = true (always)
XMLConfig: process SIP    OPTIONS  = true (always)
XMLConfig: process SIP  SUBSCRIBE  = true (always)
XMLConfig: process SIP     NOTIFY  = true (always)
XMLConfig: process SIP    PUBLISH  = true (always)
Lines in HTML = 300
Max file size for merge [MB] = 100
XMLConfig: anonymize output RTP + SIP
uname : darwin
uname : darwin
20210404_161855: Program running on OS MAC
20210404_161855: Little Endian
20210404_161855: PCAP LIB2 used
20210404_161855: uname : darwin
20210404_161856: License is activated. License valid until [14/05/2021]
20210404_161856: ------------------------------------------
20210404_161856: CMD: Start the processing...
20210404_161856: Nothing done. Use -h or --help to get some help.
20210404_161856: ##########################################
bart@Barts-Mac Downloads %
```

4.5 Fallback

No fallback.

4.6 Security Considerations

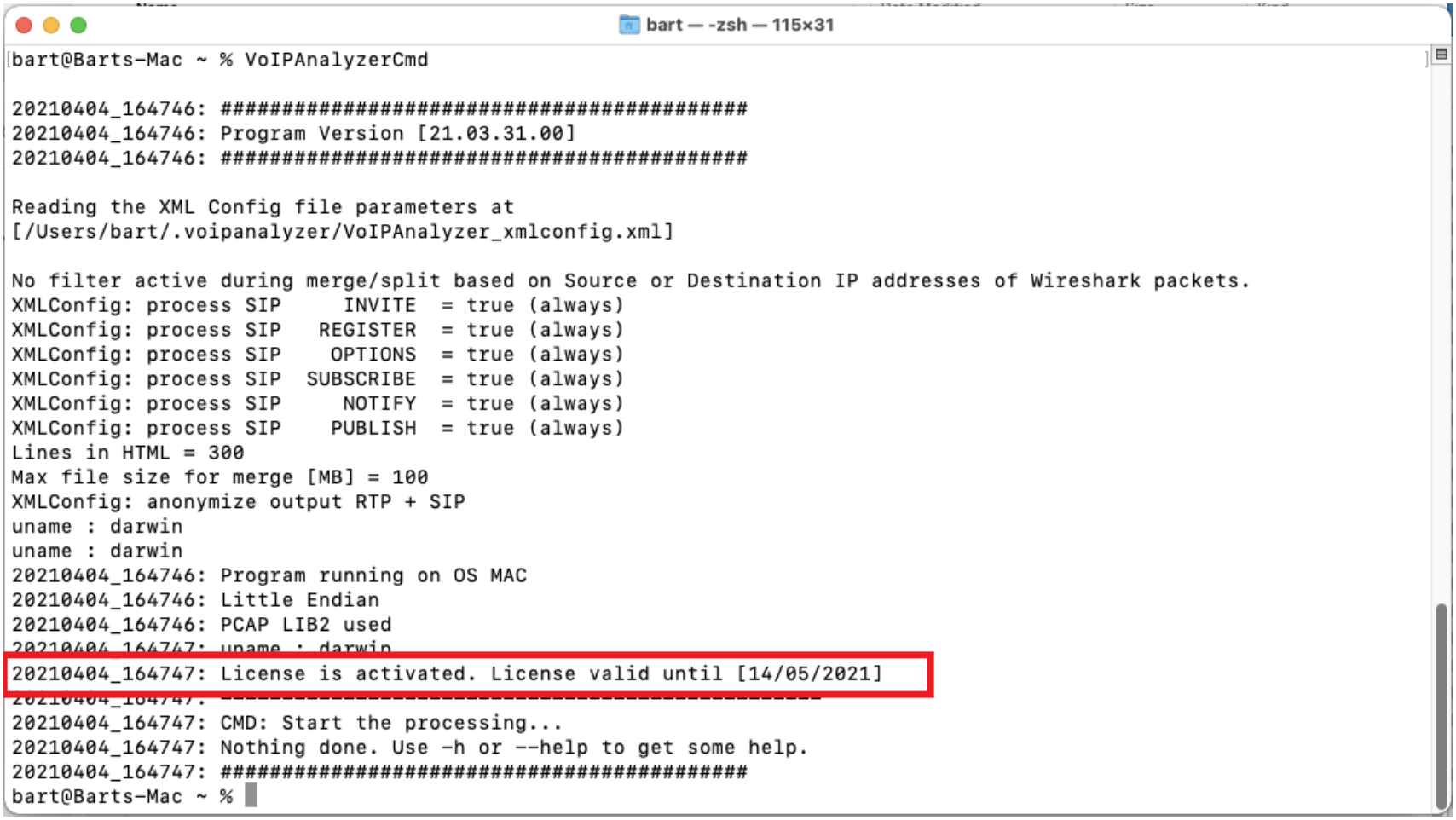
None.

4.7 Special settings and instructions

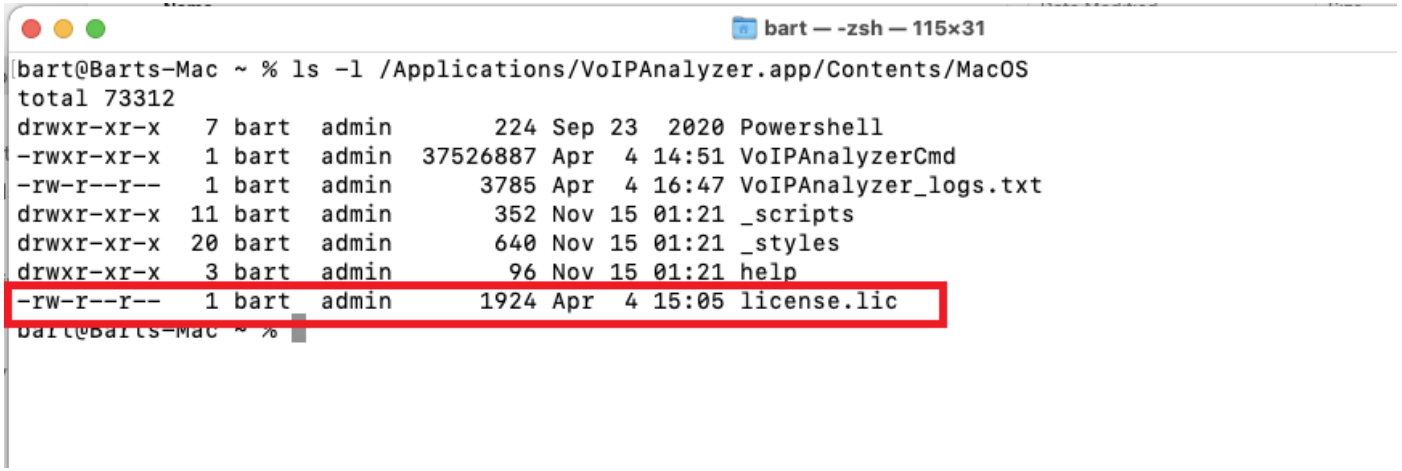
4.7.1 Configuration Requirements / Restrictions

4.7.1.1 Licensing

The license status is given when starting the VoIP Analyzer Tool from the terminal window :

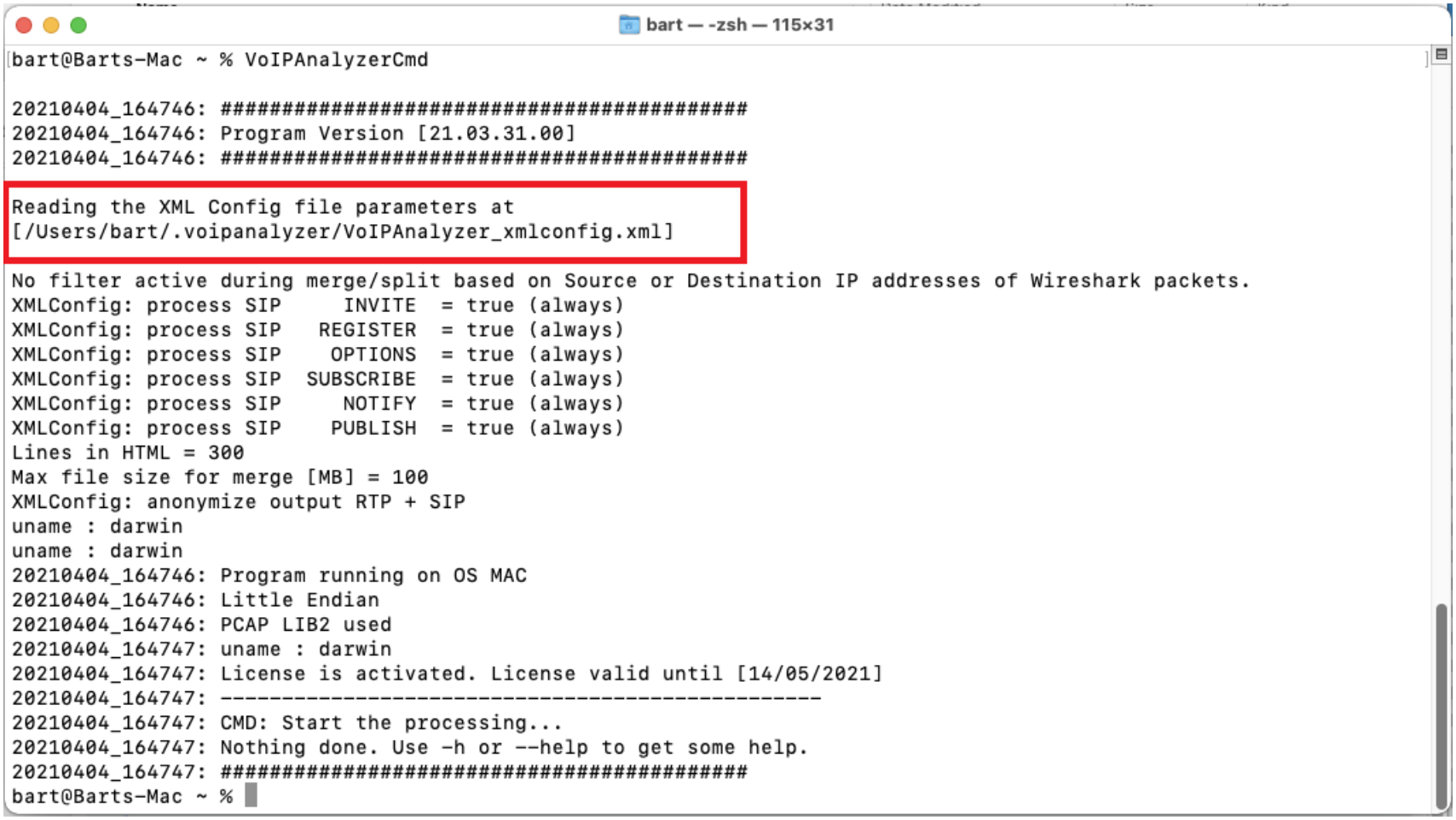


The license file is located under the /Applications/VoIPAnalyzer.app/Contents/MacOS :



4.7.1.2 XML configuration file

In the .voipanalyzer directory under the $HOME directory of the user an xml configuration file VoIPAnalyzer_xmlconfig.xml is present :



Normally nothing has to be changed in this file but some comment can be found inside :

```
-<VoIPAnalyzer>
  <Version>1.3</Version>
 -<!--
    - - -System part: general system specific parameters
  -->
 -<System>
    <DefaultStartupDirectory>/</DefaultStartupDirectory>
    <EmbedScriptsInHtml>false</EmbedScriptsInHtml>
    <EmbedStylesInHtml>false</EmbedStylesInHtml>
    <UseMinimizedScriptsInHtml>true</UseMinimizedScriptsInHtml>
    <UseMinimizedStylesInHtml>true</UseMinimizedStylesInHtml>
    <StartHtmlOverviewAfterAudit>true</StartHtmlOverviewAfterAudit>
    <OpenFileExplorerAfterAnonymize>true</OpenFileExplorerAfterAnonymize>
    <OpenFileExplorerAfterMerge>true</OpenFileExplorerAfterMerge>
    <NumberOfLinesInHtml>300</NumberOfLinesInHtml>
    <UseExcelFilteringInHtml>true</UseExcelFilteringInHtml>
  </System>
 -<Merge>
    <MergeOnlySIP_Enabled>false</MergeOnlySIP_Enabled>
    <MergeOnlyDate_Enabled>false</MergeOnlyDate_Enabled>
    <MergeOnlyDate>1/4/2020</MergeOnlyDate>
    <MergeOnlyTime_Enabled>false</MergeOnlyTime_Enabled>
    <MergeOnlyStartTime>14:00</MergeOnlyStartTime>
    <MergeOnlyStopTime>15:00</MergeOnlyStopTime>
    <MergeOnlyIpAddress_Enabled>false</MergeOnlyIpAddress_Enabled>
   -<MergeOnlyIpAddressesToFilterOnList>
     -<!--
        - - -Note: multiple entries for <MergeOnlyIpAddressesToFilterOn> are supported
      -->
     -<!--
        - - -Example: <MergeOnlyIpAddressesToFilterOn IP="192.168.56.25" Netmask="255.255.255.255">1</MergeOnlyIpAddressesToFilterOn>
      -->
      <MergeOnlyIpAddressesToFilterOn IP="192.168.56.25" Netmask="255.255.255.255">1</MergeOnlyIpAddressesToFilterOn>
      <MergeOnlyIpAddressesToFilterOn IP="10.10.10.0" Netmask="255.255.255.0">2</MergeOnlyIpAddressesToFilterOn>
    </MergeOnlyIpAddressesToFilterOnList>
    <MergeOnlySyslog_Enabled>false</MergeOnlySyslog_Enabled>
    <!-- - - -Merge file size in MB (MegaBytes) -->
    <MaxFileSizeForMergeInMB>100</MaxFileSizeForMergeInMB>
  </Merge>
 -<Audit>
    <!-- - - -SIP part: SIP related specific parameters-->
   -<SIP>
    -<PortDetect_SIP>
```

If more information is needed just email to xmlconfig@voipanalyzertool.com

4.7.2 VoIP Analyzer Tool Performance and Resilience

4.7.2.1 Vulnerabilities

It is the customer his responsibility to keep the Operating System and its components up to date.

5 Reported Problems / Symptoms under Analysis

None.

6 Restrictions, Workarounds and Hints

6.1 Restrictions

6.1.1 Restrictions or Exceptions

For the VoIPAnalyzer Tool the following restrictions apply :

- Only IPv4 addresses are supported and not IPv6 addresses.
- SIP via Transport Layer Security (TLS) is not supported.
- For macOS currently we only offer a command line, no GUI. GUI support of the tool can be supported in the future on demand.

6.2 Workarounds / Hints

6.2.1 Workarounds

None.

6.2.2 Hints

6.2.2.1 XML configuration

SIP is detected on port 5060 (and 5061 if already decoded), but also on other "known" non-default SIP ports like 5070, 10000 and 50000-50039 supported.

When other non-default SIP ports are used please add port in the xml configuration file VoIPAnalyzer_xmlconfig.xml.

Add an item such as :

<PortDetect_SIPItem>5150</PortDetect_SIPItem>

The current ports and port ranges detected are :

5060-5061

5150

5161

7777

10000

50000-50039

50070

62530-62531

65060-65061

7 Service information

7.1 License management

NA

7.1.1 Open Source Software

The product contains Open Source Software developed by third parties and licensed under an Open Source Software License. Such software files are protected by copyright. Your rights to use the Open Source Software beyond the VoIP Analyzer Tool application are governed by the relevant Open Source Software license conditions.

Your compliance with such license conditions will entitle you to use the Open Source Software as foreseen in the relevant license. In the event of conflicts between Stacked Brains license conditions and the Open Source Software license conditions, the Open Source Software conditions shall prevail with respect to the Open Source Software portions of the software.

Open Source Software is licensed free of royalties, i.e. no fees are charged for exercising the licensed rights, whereas fees may be charged for reimbursement of costs incurred by Stacked Brains to provide or distribute the software. The license conditions can be found in the source code or at the local management user interface, Utilities menu, Licenses section.

If programs embedded with the VoIP Analyzer Tool product are licensed under the GPL or LGPL conditions and these programs are not already delivered in source code form together with the VoIP Analyzer Tool, you can request the corresponding source code from Stacked Brains by contacting your sales representative.

7.2 Warranty

Stacked Brains gives no warranty for Open Source Software programs embedded with the VoIP Analyzer Tool if such programs are used in any manner other than originally intended by Stacked Brains. In such cases, warranty, if any, shall be provided by the authors of the Open Source Software as described in the license conditions referred above.

Stacked Brains specifically disclaims any warranties for defects caused by altering any Open Source Software program or its application or configuration within the VoIP Analyzer Tool. Technical support, if any, will only be provided for unmodified software.

You have no warranty claims against Stacked Brains in the event that the Open Source Software infringes the intellectual property rights of a third party.

7.3 Remote serviceability

In case remote support is wanted, the customer needs to deliver the remote access.