voip links

VoIP protokollok

SIP

- SIP - Wikipédia
- [http://www.tech-invite.com SIP Call Flow](http://www.tech-invite.com)
- SIP
- IANA Session Initiation Protocol (SIP) Parameters
- SIP RFC's
- IETF SIP RFC's

Történelem

- SIP történelem
- SER/OPENSER/KAMAILIO/SIP-ROUTER történelem

MD5 Digest

- http digest számoló eszköz
- [http://en.wikipedia.org/wiki/Digest_access_authentication]

SIP is not Simple

- The following graphs illustrate the enormus growing rate of RFCs which are related to SIP or VoIP in general.

SIP Auto answer / Page

*sip implementors *Requesting Answering Modes for the Session Initiation Protocol (SIP)

SIP Stacks

- pjsip SIP stack
- python based SIP stack
- Sofia SIP stack
- SIPit SIP stack együttműködési tesztek összefoglalói
- SIP SIMPLE client is a Software Development Kit (Python)

SIP over IPv6

- Z1FvEKmFzAsC&printsecfrontcover Voice Over IPv6

SIP-B

- SIP-B + Deverto
- SIP-B cikk
- SIP-B lejárt draft
- SIP-B definíció

Roadmaps

- Cisco SIP Roadmap
- Asterisk roadmap

SIP Identity

- draft sip-privacy-04 (Remote-Party-ID) lejárt leváltotta RFC 3323-3325
- RFC 3323 A Privacy Mechanism for the Session Initiation Protocol (SIP)
- RFC 3324 Short Term Requirements for Network Asserted Identity
- RFC 3325 (Private Extensions to the SIP for Asserted Identity within Trusted Networks)
- RFC 4474 (Enhancements for Authenticated Identity Management in the SIP)
- RFC 4474 Concerns (Tel uri + RFC 4474)
- draft-wing-sip-identity-media-02.txt

SIP Peeering ViPR (Verification involving PSTN Reachability)

- http://www.ietf.org/mail-archive/web/dispatch/current/msg00901.html
- http://tools.ietf.org/html/draft-rosenberg-dispatch-vipr-overview-02

SIP SAML

- SIP SAML Profile and Binding
- RFC 4484 Trait-Based Authorization Requirements for the Session Initiation Protocol (SIP)

SIP RACE Condition

- RFC5407 Example Call Flows of Race Conditions in SIP (Best Current Practice)

RFC

- RFC 4244 SIP History-Info header
- RFC 3842 Message Waiting Indication Event Package (pl. Asterisk voicemail is ezt használja)
- RFC 3420 sipfrag (SIP üzenetrészlet)
- RFC3303 Middlebox communication architecture and framework
- RFC3323 SIP Privacy
- Connected Identity RFC 4916
- Location Conveyance for the Session Initiation Protocol
- RFC 4458 / SIP URIs for Applications such as Voicemail and Interactive Voice Response (IVR)
- RFC3428: SIP Extension for Instant Messaging
- RFC4028 SIP Session Timer
- RFC4475 SIP Torture messages
- RFC4796 The Session Description Protocol (SDP) Content Attribute (SDP) Content Attribute / sip+content (douvideo vagy H.239 sip-en)

RAI Workgroup

- Session PEERing for Multimedia INTerconnect (SPEERMINT)

SIP Kliensek

SIP Hard kliensek

- Provisioning Parameters for Extension Mobility on Cisco SPA5x5 IP Phones
- Cisco Small Business IP Telephony Provisioning Guide
- Linksys SPA telefonok reboot reason code -ok listája
- SPA Phone Firmware and Tools fordítások *HUP fórum Linksys SPA magyarítás *Config Guide: SPA9000 Voice System Administration Guide (Release 6.1) *SPA5xx-7-x_Update_Jan-20-2010.pdf *Interoperability information for Asterisk(R)-based Phone Systems *spa resync "C" source code *Cisco/Linksys SPA50XG ringtone converter win utility *Cisco/Linksys SPA custom language XML support *SPA900 Phones XML Dictionaries, v6.1. Two new languages supported: Bulgarian and Norwegian. Includes following languages: Bulgarian (bg), Czech (cz), Danish(dk), German (de), English (en), French (fr), Croatian (hr), Hungarian (hu), Italian (it), Dutch (nl), Norwegian (no), Polish (pl), Portuguese (pt), Romainan (ro), Russian (ru), Spanish (es), Swedish (se), Slovak (sk), Slovenian (sl), Turkish (tu).
- Sipura/Linksys/Cisco SPA 5xx/9xx LDAP Corporate Directory
- SPA900_LDAP_CORP_DIR_V11.pdf
- Cisco 79xx/SPA5xx XML directory
- XML phone apps SPA5xx
- CiscoIPPhone object supported by SPA 5xx series IP Phones
- Support for SIP NOTIFY XML-Service event on SPA500 IP Phones
- XML Services for SPA500 series IP Phone documentation
- Small Business IP Phone Documentation
- Cisco SPA5xxG IP Phones and DHCP OPTIONS
- SPA 50X XML Dictionaries v7.4.X

Fogyasztás

- ip telefonok fogyasztásának összehasonlítása

SIP Soft kliensek

- qutecom

- portsip pangolin
- x-lite / eye-beam / bria
- twinkle
- Blink MacOSX
- phonerlite windows (ipv6!)
- mercuro windows / windows mobile
- minisip
- Enigma windows (XP/Vista/7) / Linux (Fedora)
- linphone linux
- pjsua console sip kliens

SIP Mobil kliensek

- Android + SIP = sipdroid

SIP Trunking

- What is a SIP Trunk Anyway?
- Representing Trunk Groups in tel/sip Uniform Resource Identifiers (URIs)
- Trunk Group Use in ENUM
- OPENSER/Kamailio Devel Guide

SIP Redundancy / HA

- SIP_failover_and_redundancy.pdf

SIP Troubleshooting tools

- http://cvs.berlios.de/cgi-bin/viewcvs.cgi/ser/sip_router/utils/sipgrep/
- http://www.wireshark.org/ wireshark -k -i <(ssh 192.168.1.1 "sudo tshark -w - -i eth0")
- tshark (console wireshark) tshark -i eth0 -t ad -R 'sip'-S -w /tmp/sip.cap

H.323

- H.323 Wikipedia

- H.323 paketizer
- H.323 rövid mini tutorial magyarul

SIP vs. H.323

- http://www.packetizer.com/voip/h323_vs_sip/
- http://hive.packetizer.com/users/h323forum/papers/Service_Architecures_SIP-H323.pdf
- SIP and H.323

Open Source hívásirányítók

- VoIP SoftSwitchek PBX-ek gyűjteménye..
- www.asterisk.org
- www.kamailio.org
- www.opensips.org
- sip-router.org project (common collaboration framework to X-SER)
- www.ser.org
- www.gnugk.org (csak H.323)
- openimscore.org opensource IMS

SIP-Router

- Kamailio 3.0.x - DYK - ToC

"copy left" / "public domain" / "Creative Common" Music on Hold zenék

- Wikipedia ogg file-ok
- opsound.org mp3 file-ok
- asterisk Creative Common
- http://blogs.digium.com/2009/08/18/asterisk-music-on-hold-changes/

Location / Végpontok azonosítása / Azonosító gazdálkodás

- országhívó kódok wikipedia country codes

Magyar Azonosító gazdálkodás

- Azonosítógazdálkodással és számhordozhatósággal kapcsolatos jogszabályok
- 164/2005. (VIII. 16.) Korm. rendelet az elektronikus hírközlő hálózatok azonosítóinak nemzeti felosztási tervéről (rövid számok!!!)
- Magyar Nemzeti számozási terv
- www.nhh.hu Azonosító gazdálkodás

Külföldi Azonosító gazdálkodás

- NANPA: North American Numbering Plan Administration
- voip-info.org +3883 - European Telephony Numbering Space - ETNS
- +3883 - European Telephony Numbering Space - ETNS
- iNum is making use of the +883 global country code newly created by the ITU (International Telecommunication Union).

ENUM

- NREN ENUM Az európai kutató hálózatok privát ENUM fája.
- ENUM Wikipédia
- RIPE ENUM
- Magyar ENUM pilot
- Enum fák állapota
- IANA Enumservice Registrations
- GARR NRENUM directory

RFC

- The E.164 to Uniform Resource Identifiers (URI) Dynamic Delegation Discovery System (DDDS) Application (ENUM)
- RFC 3764 enumservice registration for Session Initiation Protocol (SIP) "E2U+SIP" enumservice for SIP
- RFC 3263 - Session Initiation Protocol (SIP): Locating SIP Servers (D2X)
- SIP E.164 Problem Statement

ENUM tutorial

- Japán tutorial (2003)
- AT&T (2003)

TRIP/ITAD

- ITAD Subscriber Numbers SIP.edu
- TRIP tutorial
- A Framework for Telephony Routing over IP RFC2871
- Telephony Routing over IP (TRIP) RFC3219
- TRIP Cisco IOS 12.3

DUNDi

- DUNDi website
- DUNDi draft

Test Framework-ök (Benchmark,Compatibility,Performance,Qos,MOS,PESQ,r-factror,E-modell,VQmon,Testnumber...)

- pbx-test Asteriskhez
- SIPp
- SFTF SIP Foudry Test Framework
- SIPSTONE
- VoIP QOS Mos mérő:testyourvoip/Brix networks
- VoIP QOS Mos mérő: Test My Connection for VoIP Quality/VisualWare
- VoIP Osztrák Tesztszámok
- voip-info.org Call Quality Metrics (PESQ, MOS)
- Cisco IOS IP SLA Configuration Guide
- Cisco IOS IP SLAs Command Reference
- The Open Speech Repository (Freely usable speech files in multiple languages for use in Voice over IP testing)
- voice quality measurement
- Online Diagnostic Tools for Network Managers
- Telchemy VQmon (Technical Reference)
- miTester for SIP
- sip-dig dns SRV NAPTR tester
- IPv6 Ready
- E-model R-Factor MoS
- E-model R-Factor MoS ITU calculator
- E-model tutorial
- RFC4475 SIP Torture Test Messages SIPit
- media-loopback An Extension to the Session Description Protocol (SDP) for Media Loopback draft-ietf-mmusic-media-loopback-13 (?Cisco/Linksys/SIPURA SPA?)
- loopback testing
- Cisco Enterprise QoS Solution Reference Network Design Guide Wideband Demo SIP URI G.711 és G.722 között lehet váltogatni a "#" megnyomásával
- sip:wbdemo@conf.zipdx.com vagy sip:wbdemo@zipdx.com
- sip-viewer

VoIP biztonság / Security

- VoIP Crack
- cain password Cain & Abel is a password recovery tool for Microsoft Operating Systems.

- IETF Draft Domain Certificates in SIP SubjectAltName használata
- IETF Draft SIP + X.509 Extended Key Usage
- VOIPSA.org voip security resources
- SIP overload control problem statement
- SIP DoS
- SIP RTP DoS
- Overview of SIP Media Security Options - Dan Wing
- sipvicious /sip scanner tool/

Videokonferencia és VoIP Prezentációk / Konferenciák / Portálok / Tutorial / Szerveztek

Tutorial

- SIP Pocket Guide
- SIP.edu cookbook
- SIP mini overview from Cisco
- A Hitchhiker's Guide to the Session Initiation Protocol (SIP)
- Tekelec Webinars
- TEKELEC SIP Tutorial 1
- TEKELEC SIP Tutorial 2
- TEKELEC SIP Tutorial 3
- TEKELEC SIP Tutorial 4
- Upgrading the Next-generation Network Part II - Layer 5 Core SIP Routing
- OpenSer/Kamailio Admin Course 2007.08.29
- SIP tutorial (iptel.org)
- UNC realtime Architecture Christian Schlatter
- PPKE ITK Internetes médiakommunikáció VoIP
- CPL Tutorial
- opensips webinar
- opensips webinars home

- 3CX IP PBX-szel, SIP-pel és VOIP-pel kapcsolatos GYIK
- kezdolap&Almenuhogyanmukodik VoIP tudástár magyarul, kezdőknek
- OpenSIPS loadbalancing példa
- Trixbox telepítés
- XCAP tutorial
- Overview of SIP - Cisco
- Bilicki Vilmos Szegedi Egyetem VoIP - SIP - XMPP
- Kamailio Knowledge Base (kb.aspito.com)
- VoIP in-depth: An introduction to the SIP protocol, Part 1

Asterisk Books Tutorial

- O'Reilly Book Asterisk: The Future of Telephony, 2nd Edition
- Asterisk: The Future of Telephony html formában
- Asterisk Tutorial
- VIM + Asterisk extension.conf syntax highlighting
- Asterisk SIP User vs. Peer

Konferenciák

- http://www.cluecon.com/
- http://www.von.com/
- http://www.astricon.net/
- http://www.amoocon.de/
- NETWORKERS 2004
- OpenSIPS Amoocon 2009

Portálok

- Wainhouse Research Videoconference Portal analyzes the market trends, technologies/products, vendors, applications, and related services in the Unified Communications and rich media conferencing fields.
- Tárcsahang

Szervezetek

- sipfórum
- Terena: Enhanced Communication Services - Task Force

PSTN / POTS

- FXS vs. FXO
- ISDN E1 failover switch

Cisco ISDN gateway configuration

- Cisco DSP calculator (jogosultság szükséges a használatához)
- Voice Hardware Compatibility Matrix (Cisco 17/26/28/36/37/38xx, VG200, Catalyst 4500/4000, Catalyst 6xxx)
- Table 29 Dial-Peer Matching Rules for Inbound URI in SIP Calls
- Table 16 Dial-Peer Matching Rules for Outbound URI
- Cisco IOS Voice Command Reference
- Cisco IOS Fax, Modem, and Text Support over IP Configuration Guide, Release 12.4
- Understanding Inbound and Outbound Dial Peers Matching on IOS Platforms
- Cisco IOS Voice Port Configuration Guide (Book Length PDF)
- Fax over IP T.37 Store and Forward Fax (Nagyvonalú leírás és részletes minta példa. Támogatott e-mail formátum részletezése
- VoIP Gateway Trunk and Carrier Based Routing Enhancements
- Radius Accounting Cisco VSA

Troubleshooting

- Cisco Telephony Interface Architecture felépítés
- Voice Call Debug Filtering on Cisco Voice Gateways
- SIP Debug Output Filtering
- Cisco WIKI: Cisco IOS Voice Troubleshooting and Monitoring
- Cisco Fax hibakeresés
- Cisco Voice Command Reference
- Session Initiation Protocol Gateway Call Flows

- Cisco IOS Voice Troubleshooting and Monitoring Guide, Release 12.4
- 12.4T Configuration Guide
- Understanding debug isdn q931 Disconnect Cause Codes
- Understanding the show controllers E1 Command / E1 hibakeresés
- Voice Translation Rules / test voice translation-rule

NAT STUN / TURN / ICE , XTUNNEL

- Interactive Connection Establishment /ICE/ - Rosenberg (NAT Traversal)
- Xtunnel a counterpath/xten eyebeam/bria/x-lite termékéhez
- NAT traversal ICE/TURN/STUN
- turnserver TURN implementáció
- ICE STUN TURN tutorial
- ICE: the ultimate way of beating NAT in SIP (amoocon)

Codec, hang és video kódolás

- Digital Pictures Compression jegyzet
- Hang és Video kodekek listája
- Polycom voice codec white paper
- H.264 SVC
- Media formátumok és a konténerek

Directory, H.350

- H.350 Cookbook
- H.350 IETF RFC 3944

Speciális Hangjelzések

- speciális hangjelzések(Special Information Tone = SIT)

Provisioning

- A Framework for Session Initiation Protocol User Agent Profile Delivery (configuration provisioning)

DHCP

- DHCP opciók
- dhcp option sip-server /Dynamic Host Configuration Protocol (DHCP-for-IPv4) Option for Session Initiation Protocol (SIP) Servers/
- Cisco dhcp option 150 vs.66
- Cisco support forum dhcp option 150 or 66

QoS

- RTCP XR
- Assessing VoIP Call Quality Using the E-model

Szolgáltatók

- http://voipszolgaltato.lap.hu/

Videokonferencia

- Tandberg SIP támogatás
- TANDBERG University: Introduction to TANDBERG and Video Conferencing
- Videoconference Manual

Vertical Service Code

- GUIDELINES FOR IMPLEMENTATION SUPPLEMENTARY SERVICE CODES
- MAN MACHINE INTERFACE (MMI)

- Vertical service code
- Vertical Service Codes / Code Definitions

Példa nevek

- Alice Bob Carol Dave ... példa nevek A,B,C,D

További érdekes helyek

- voip-info.org (Open Source VoIP Biblia)
- Wikipedia VoIP
- WeSIP (adds a J2EE layer to OpenSER)
- XCAP Server Implementáció
- OpenSIPS Control Panel
- The SIPconnect Technical Recommendation *White Paper on SIP Servers (RADVISION)
- OpenSER config generator sip:wizard
- OpenXCAP python XCAP implementáció
- A list of interesting publications that are related to the Sip-router project, SIP servers, or SIP and related protocols in general.
- cisco UC poster
- SIP NAT Best Practices
- SIP-to-SIP Connections on a Cisco Unified Border Element (CUBE)
- Cisco Unified Border Element Configuration Guide
- Troubleshooting and Debugging VoIP Call Basics hogyan működik az IOS képek
- fényképek végpontok/ATA-k/stb.
- SIP problems: Actions Addressing Identified Issues with the Session Initiation Protocol's (SIP) Non-INVITE Transaction RFC4320
- SIP problems: Problems Identified Associated with the Session Initiation Protocol's (SIP) Non-INVITE Transaction RFC4321
- Dial Peer Configuration on Voice Gateway Routers
- Cisco Unified Survivable Remote Site Telephony Configuration Guides SRST
- RFC3263 DNS feloldás Best Current Practice
- Cisco telefon XML API 5.1
- Cisco telefon XML API 7.1

- 10.1.1.148.8045&reprep1&type=pdf SIP Proxy Server Effectiveness Master's Thesis Jan Jan´ak
- GEANT3 Campus Best Practice (fények hang stb. videokonferencia terem tervezéshez)
- H.264/SVC demo Global IP Solutions
- icemission softphone
- ITU Local call progress tones2003
- Local call progress tones2010
- Hungary (hun) Phone Settings and Frequencies
- PROTOS sip test framework
- World Telephone Numbering Guide
- CESNET VoIP Security
- CESNET VoIP
- DNS ZoneCheck tool
- cisco 15.1 IOS toll fraud prevention
- Universal Personal Telecommunications
- SIP vs XMPP or SIP and XMPP

Változat #1
document-uploader hozta létre 2025-08-07 12:00:03 CEST
document-uploader frissítette 2025-08-07 12:00:03 CEST