VoIP mit Asterisk PBX

Voice over IP Telephonie mit Asterisk Nebenstellenanlagen

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VoIP

- Technologiesituation
- Netzprotokolle
- Asterisk PBX-Konfiguration
- Soft/ATA Klient-Konfiguration
- Demo

Technologiesituation

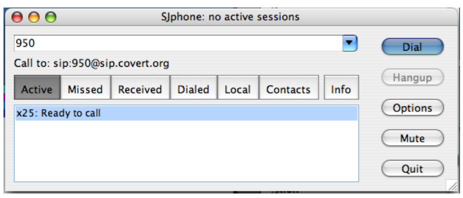
- VoIP am Anfang: intern, auf privater Infrastruktur
- Jetzt: im öffentlichen Internet
- Hunderte von Providern
- PC <--> PC, PC <--> PSTN
- ATA (analog telephone adapter)

Softphone

- Telephonieren vom PC
- Viele Produkte, auch kostenlose

X-Lite

SJphone



Analog Adapter

- Cisco ATA-186
- Sipura (tiptel in Deutschland)
- HandyTone
- FRITZ!box
- GrandStream (VoIP Telephone)

Voice over IP auf dem Vormarsch

- **Jeder siebte Deutsche will per Internet telefonieren**
- **Schon acht von zehn Deutschen kennen Internet-Telefonie**
- **44 Prozent begeistert von Einsatzmöglichkeiten**
- **14 Prozent planen Einstieg binnen zwölf Monaten**
- **Niedrige Kosten und Zusatzfunktionen überzeugen**
- **Neue Studie von Roland Berger Market Research**
- **(VoIP News 4.4.2005)**
- **Gute Infoquelle: http://www.ip-phone-forum.de/**

Anbieter für Deutschland

- AdvanceCall
- babble.net
- BlueSIP
- Broadnet
- dus.net
- freenet iPhone
- GMX NetPhone
- Nikotel
- PURtel
- Sipgate
- Sipsnip
- WEB.DE FreePhone

Sipgate.de

- Kostenlose verteilung geographischer bzw 01801 Nummer
- Portierung ca. 4. Quartal 2005
- Softphone, Analog Adapter, Asterisk
- Kostenfrei zu anderen VoIP Nummer
- Verkauft ggf Analog Adapter
- Pre-pay
- Billige Minutenpreise

Einstieg

- Softphone downloaden, ggf. ATA
- Bei einem Anbieter kostenlos anmelden (D-PSTN Rufnummer)
- Kostenlose Seattle(Umgebung) Rfnr
- Ankommende Gespräche kostenfrei
- Gespräche an VoIP-Teilnehmer auch
- Konto aufladen: ins PSTN anrufen

Bequemer Telefonieren

- ATA oder VoIP kaufen
- Normale Telefone anschließen
- ggf VoIP Nebenstellenanlage (OpenSource Asterisk) einsetzen
- Bei mehreren Providern anmelden
- Least-cost Routing in Asterisk

Kostenlos in die USA

- Kostenlose SIP-Adresse (Provider)
- Kostenlose 360 Area Code Nr bei IPKall.com auf die SIP-Adresse
- Amazon Yellow Pages (yp.a9.com)
  - Firma in den Yellow Pages suchen
  - 360-Nummer klingelt hier in Deutschland
  - Verbindung geht dann an die Firma
  - Hilton Disney World http://www.amazon.com/gp/yp/B0004UDK2G/002-8827715-4872835

Netwerkprotokolle

- SIP - Session Initiation Protocol
  RFC 3261
- RTP/RTCP - Protocol for Real-Time
  RFC 1889
- STUN - Simple Traversal UDP NAT
  RFC 3489
- Auch H.323, IAX, RFC2833(DTMF), ...

SIP - Session Initiation

Anruf machen:

-> INVITE

<- SIP 407 Proxy-Authentication-Required

-> ACK

-> INVITE mit Authentifizierung (MD5)

<- SIP 100 Trying

<- SIP 183 Session Progress

<- SIP 200 OK

-> ACK

SIP - Details (1 ->)

INVITE sip:08002255288@sip.covert.org SIP/2.0

Contact: <sip:x25@62.8.199.106:5060>

Call-ID: E0AF52A6-1DD1-11B2-9801-F60C9CB0269B@192.168.2.1

Content-Type: application/sdp

From: "John Covert"<sip:x25@sip.covert.org>;tag=391186259149179312 5

CSeq: 1 INVITE

Max-Forwards: 70

To: <sip:08002255288@sip.covert.org>

Via: SIP/2.0/UDP
192.168.2.1;rport;branch=z9hG4bKc0a802010131c9b14253cb04297413fe
00000007

User-Agent: SJLabs-SJphone/1.30.256b

Content-Length: 366

SIP - Details (2 <-)

SIP/2.0 407 Proxy Authentication Required

Via: SIP/2.0/UDP
192.168.2.1;rport;branch=z9hG4bKc0a802010131c9b14253cb04297413fe00000007;received=62.8.199.106

From: "John Covert"<sip:x25@sip.covert.org>;tag=391186259149179 3125

To: <sip:08002255288@sip.covert.org>;tag=as28f7f493

Call-ID: E0AF52A6-1DD1-11B2-9801-F60C9CB0269B@192.168.2.1

CSeq: 1 INVITE

User-Agent: Asterisk PBX

Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER

Contact:

Proxy-Authenticate: Digest realm="asterisk", nonce="2ed9f3e8"

Content-Length: 0

SIP - Details (3 ->)

ACK sip:08002255288@sip.covert.org SIP/2.0

Content-Length: 0

Call-ID: E0AF52A6-1DD1-11B2-9801-F60C9CB0269B@192.168.2.1

Max-Forwards: 70

CSeq: 1 ACK

From: "John Covert"<sip:x25@sip.covert.org>;tag=3911862591491793125

To: <sip:08002255288@sip.covert.org>;tag=as28f7f493

Via: SIP/2.0/UDP
192.168.2.1;branch=z9hG4bKc0a802010131c9b14253cb04297413fe00000007

SIP - Details (4.0 ->)

INVITE sip:08002255288@sip.covert.org SIP/2.0

Content-Length: 366

Contact: <sip:x25@62.8.199.106:5060>

Call-ID: E0AF52A6-1DD1-11B2-9801-F60C9CB0269B@192.168.2.1

Content-Type: application/sdp

From: "John Covert"<sip:x25@sip.covert.org>;tag=3911862591491793125

CSeq: 2 INVITE

Max-Forwards: 70

To: <sip:08002255288@sip.covert.org>

Via: SIP/2.0/UDP 192.168.2.1;rport;branch=z9hG4bKc0a80201000000184253cb0403cc9df500000009

User-Agent: SJLabs-SJphone/1.30.256b

Proxy-Authorization: Digest username="x25",realm="asterisk",nonce="2ed9f3e8",uri="sip:08002255288@sip.covert.org",response="7fe198361122847084e88800817a85c3"

SIP - Details (4.1 ->)

v=0

o=- 3321776515 3321776515 IN IP4 62.8.199.106

s=SJphone

c=IN IP4 62.8.199.106

t=0 0

a=direction:active

m=audio 16384 RTP/AVP 0 8 3 97 98 110 101

a=rtpmap:0 PCMU/8000

a=rtpmap:8 PCMA/8000

a=rtpmap:3 GSM/8000

a=rtpmap:97 iLBC/8000

a=rtpmap:98 iLBC/8000

a=fmtp:98 mode=20

a=rtpmap:110 speex/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11,16

SIP - Details (5 <-)

SIP/2.0 100 Trying

Via: SIP/2.0/UDP 192.168.2.1;rport;branch=z9hG4bKc0a8020100000018 4253cb0403cc9df500000009;received=62.8.199.106

From: "John Covert"<sip:x25@sip.covert.org>;tag=391186259149179 3125

To: <sip:08002255288@sip.covert.org>;tag=as671c1c5f

Call-ID: E0AF52A6-1DD1-11B2-9801-F60C9CB0269B@192.168.2.1

CSeq: 2 INVITE

User-Agent: Asterisk PBX

Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER

Contact: <sip:08002255288@24.63.81.3>

Content-Length: 0

SIP - Details (6.0 <-)

SIP/2.0 183 Session Progress

Via: SIP/2.0/UDP 192.168.2.1;rport;branch=z9hG4bKc0a8020100000018 4253cb0403cc9df500000009;received=62.8.199.106

From: "John Covert"<sip:x25@sip.covert.org>;tag=391186259149179 3125

To: <sip:08002255288@sip.covert.org>;tag=as671c1c5f

Call-ID: E0AF52A6-1DD1-11B2-9801-F60C9CB0269B@192.168.2.1

CSeq: 2 INVITE

User-Agent: Asterisk PBX

Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER

Contact: <sip:08002255288@24.63.81.3>

Content-Type: application/sdp

Content-Length: 207

SIP - Details (6.1 <-)

v=0
o=root 3427 3427 IN IP4 24.63.81.3
s=session
c=IN IP4 24.63.81.3
t=0 0
m=audio 25604 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16

(n.b. Asterisk kann noch mehr Audioformate)

SIP - Details (7 <-)

SIP/2.0 200 OK
Via: SIP/2.0/UDP
192.168.2.1;rport;branch=z9hG4bKc0a8020100000184253cb0403cc9df500000009;received=62.8.199.106
From: "John Covert"<sip:x25@sip.covert.org>;tag=391186259149179312 5
To: <sip:08002255288@sip.covert.org>;tag=as671c1c5f
Call-ID: E0AF52A6-1DD1-11B2-9801-F60C9CB0269B@192.168.2.1
CSeq: 2 INVITE
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER
Contact: <sip:08002255288@24.63.81.3>
Content-Type: application/sdp
Content-Length: 207

[Contents sind dem vorigen Nachrichten gleich]

SIP - Details (8 ->)

ACK sip:08002255288@24.63.81.3 SIP/2.0
Content-Length: 0
Contact: <sip:x25@62.8.199.106:5060>
Call-ID: E0AF52A6-1DD1-11B2-9801-F60C9CB0269B@192.168.2.1
Max-Forwards: 70
CSeq: 2 ACK
From: <sip:x25@sip.covert.org>;tag=3911862591491793125
To: <sip:08002255288@sip.covert.org>;tag=as671c1c5f
User-Agent: SJLabs-SJphone/1.30.256b
Via: SIP/2.0/UDP 192.168.2.1;rport;branch=z9hG4bKc0a802010131c9b14253cb086e9641c50000000d

Und jetzt laufen also die RTP-Paketen.

SIP - Weitere Funktionen

- Anruf beenden
  - -> BYE
  - <- OK
- Registrieren
  - -> REGISTER
  - <- SIP Trying
  - <- SIP 407 Proxy-Authentication...

SIP - Weiteres

- Weiterleiten
  - REFER (RFC 3515)
- Ausloggen:
  - REGISTER mit Expires: 0
- Informieren
  - OPTIONS

RTP/RTCP

- Audio-Paketen
  - PCMU (µlaw)
  - PCMA (alaw)
  - GSM
  - SPEEX
  - iLBC
  - ...

STUN

- Simple Transversal of User Datagram Protocol (UDP) through Network Address Translators (NATs)
  - Ermöglicht SIP von hinter einem NAT. Nicht besonders gut wenn beiden Enden hinter NAT stecken.

Weitere Protokolle

- H.323 - zumeist von SIP ersetzt
- IAX - Inter-Asterisk-Exchange
  - Bevorzugt von meisten Asterisk-Hacker. Bessere NAT-Traversal.
  - Nicht standardisiert.

ASTERISK PBX

- Open Source
- Linux, MacOS/X, ...
- Digium.com - hardware
- FXO (office: AMTsverbindung)
- FXS (station: NSverbindung)
- Hardware am PBX unnötig

Asterisk Features

- Konfigurierbar, Programierbar
- Voicemail - Emailen abg. Nachr.
- Menüs
- Ansagen
- AGI (Application Generic Intfc)
  - Irgendetwas (z.B. DB-Nachfrage)

Asterisk Konfigurieren

- Runterladen, CVS, package.
- (MacOS Installer doppelklicken)
- Zwei Dateien editieren bzw. „Asterisk-at-Home“ Konfigurator oder Sunshine Mac Konfigurator aufrufen.
- Basta

Asterisk config sip.conf

- Google asterisk config sip.conf

Meine Anbieter-Einträge für sipgate:

```
register => 7stelligeNr:SIPPasswort@sipgate.de/nnnnextconf

[sipgate-b]
type=friend
secret=passwort
username=7stelligeNr
fromuser=7stelligeNr
fromdomain=sipgate.de
host=sipgate.de
dtmfmode=rfc2833
nat=no
insecure=very
canreinvite=no  ; yes wäre vielleicht ok, da mein Asterisk nicht mehr hinter
context=inbound-sipgate                        ; NAT steckt.
```

Asterisk sip.conf NS

- Eine Nebenstelle am Cisco ATA

```
[x26]
type=friend
secret=password
nat=yes
host=dynamic
canreinvite=no          ; Vielleicht mit dem neusten firmware „yes“?
dtmfmode=rfc2833
qualify=200             ; Qualify peer is no more than 200ms away
context=dialstation26
callerid=Cisco 26 <26>
```

Asterisk config extensions.conf

- Ankommende sipgate-Anrufe

```
RINGALL=SIP/x21&SIP/x22&SIP/x23&SIP/x26
[inbound-sipgate]
exten => nnnnn,1,Dial(${RINGALL},120,t)
exten => nnnnn,2,Macro(fastbusy)
exten => h,1,Hangup
```

Asterisk (2) extensions.conf

- Anrufe von x26

```
[dialstation26]
exten => _.,1,SetVar(ThisExt=26)
exten => _.,2,Gotoif($[${EXTEN} = 26]
   ?busy,s,1:dialstation,${EXTEN},1)
exten => h,1,Hangup
[busy]
exten => s,1,Playtones(busy)
exten => s,2,Busy
exten => h,1,Hangup
```

Asterisk (3) extensions.conf

```
[dialstation]
include => babble-out
include => sipgate-out
exten => _2X,1,Dial(SIP/x${EXTEN},120)
exten => _2X,1,Macro(fastbusy)
exten => 112,1,Dial(SIP/112@tiptel-pstnport,120)
exten => 112,1,Macro(fastbusy)
exten => 811,1,Goto(ringback,s,1)
[sipgate-out] ; Beispiel für Berliner
exten => _0.,1,Dial(SIP/${EXTEN}@sipgate-b,120)
exten => _0.,2,Macro(fastbusy)
exten => _[2-9].,1,Dial(SIP/030${EXTEN}@sipgate-b,120)
exten => _[2-9].,2,Macro(fastbusy)
```

Asterisk (4) extensions.conf

```
[babble-out]
exten => _0044.,1,Dial(SIP/${EXTEN}@babble,120)
exten => _0044.,2,Macro(fastbusy)

; weitere „Babble“-Länder auch eintragen

[fwd-800-nr]
exten => _001800.,1,Dial(SIP/${EXTEN:2}@fwd.pulver.com...
... auch 866,877,888, und dazu noch
Norwegen (_0047800.), NL (_0031800.), Japan (_00810120.)
```

Asterisk Menü

```
[reachedmypbx]
exten => s,1,Wait(1)                    ; Wait a second, just for fun
exten => s,2,Answer(${CALLERID})        ; Answer the line
exten => s,3,Wait(1)                    ; Wait another second for SIP delay
exten => s,4,DigitTimeout,5             ; Set Digit Timeout to 5 seconds
exten => s,5,ResponseTimeout,10  ; Set Response Timeout to 10 seconds
exten => s,6,Setvar(c=1)
exten => s,7,BackGround(/Users/jcovert/asterisk/reachedmypbx)

include => localstations

exten => 84,1,SayUnixTime(,,R)
exten => 84,2,Macro(loopcheck)
exten => 84,3,Goto(s,7)

exten => 5646,1,Goto(inbound-ringall,999,1)    ; John
exten => 726,1,Goto(inbound-ringall,997,1)     ; Pam

exten => i,1,Macro(loopcheck)
exten => i,2,Playback(invalid)
exten => i,3,Goto(s,7)
exten => t,1,Goto(inbound-ringall,999,1)
exten => t,2,Hangup
exten => h,1,Hangup
```

Asterisk Ringprobe

Emulierung einer No. 1 Xbar, C&P Telco, circa 1960

```
[ringback] ; 811-dialtone-6.  extension readback on zero.
exten => s,1,Playtones(420+520)
exten => s,2,SetVar(hup=0)
exten => i,1,Hangup
exten => 6,1,Playtones(240+260)
exten => 6,2,SetVar(hup=1)
exten => 6,3,Wait(30)
exten => 6,4,SetVar(hup=0)
exten => 6,5,Hangup
exten => 0,1,StopPlaytones
exten => 0,2,Wait(1)
exten => 0,3,SayNumber(${ThisExt}) ; ok for 2D - longer would
    require SayDigits
exten => 0,4,Goto(s,1)
```

Asterisk Ringprobe (2)

```
exten => h,1,Gotoif(${hup}?2:6)
exten => h,2,SetVar(ep=${EPOCH})
exten => h,3,System(printf "Channel:
    SIP/x${ThisExt}\nContext:
    dialstation${ThisExt}\nExtension: 811\nCallerID:
    ${ThisExt}" >/var/spool/asterisk/ringback${ThisExt}-${ep})
exten => h,4,System(sleep 1)
exten => h,5,System(mv
    /var/spool/asterisk/ringback${ThisExt}-${ep}
    /var/spool/asterisk/outgoing/ringback${ThisExt}-${ep})
exten => h,6,Hangup
```

Meine Konfiguration

```
Asterisk> sip show peers
Name/username     Host
x27/x27           62.8.199.106
x26/x26           62.8.199.106
x25/x25           (Unspecified)
x24               (Unspecified)
x23/x23           192.168.0.9
x22/x22           192.168.0.8
x21/x21           192.168.0.8
faktortel/09500   202.125.42.141
stanaphone/5162   204.147.183.18
squillo/jcovert   194.177.124.179
msgntit/5301044   212.97.59.76
e164org/jcovert   204.209.140.71
von-g/1678.....   216.115.25.198
mlg5433/x5433     63.67.19.122
wacker/5433       207.224.49.209
simpletelecom/s   63.218.92.199
babble/jcovert    62.73.169.12
sipgate-b/.....   217.10.79.9
fwd1/......       69.90.155.70
iptel/........    195.37.77.99
pstn5433/pstn54   192.168.0.9
jrcsipintoa1/jr   12.40.50.254
```

Softphone config

- X-Lite auf Asterisk

  Menu | System Settings | SIP Proxy | default
  Display Name: Full Name
  User Name & Authorization User: <username>
  Password: <secret>
  Domain/Realm: x.x.x.x
  SIP Proxy: x.x.x.x
  STUN: default (benutz den STUN-server bei xten.com)
  x.x.x.x = IP address of Asterisk

- http://www.voip-info.org/wiki-Asterisk+phone+xten+xlite

ATA config

- Cisco-Beispiel

DHCP=1;
UID0=x26; PWD0=secret; UID1=x27; PWD0=secret;
SipRegOn=1; GkOrProxy=x.x.x.x; OutBoundProxy=x.x.x.x
NATserver=stun.fwdnet.net:3478
Vielleicht SIPport auf 5062 Ändern, konflikt zu vermeiden.
AudioMode: **0x00140014**

Demo und Fragen

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