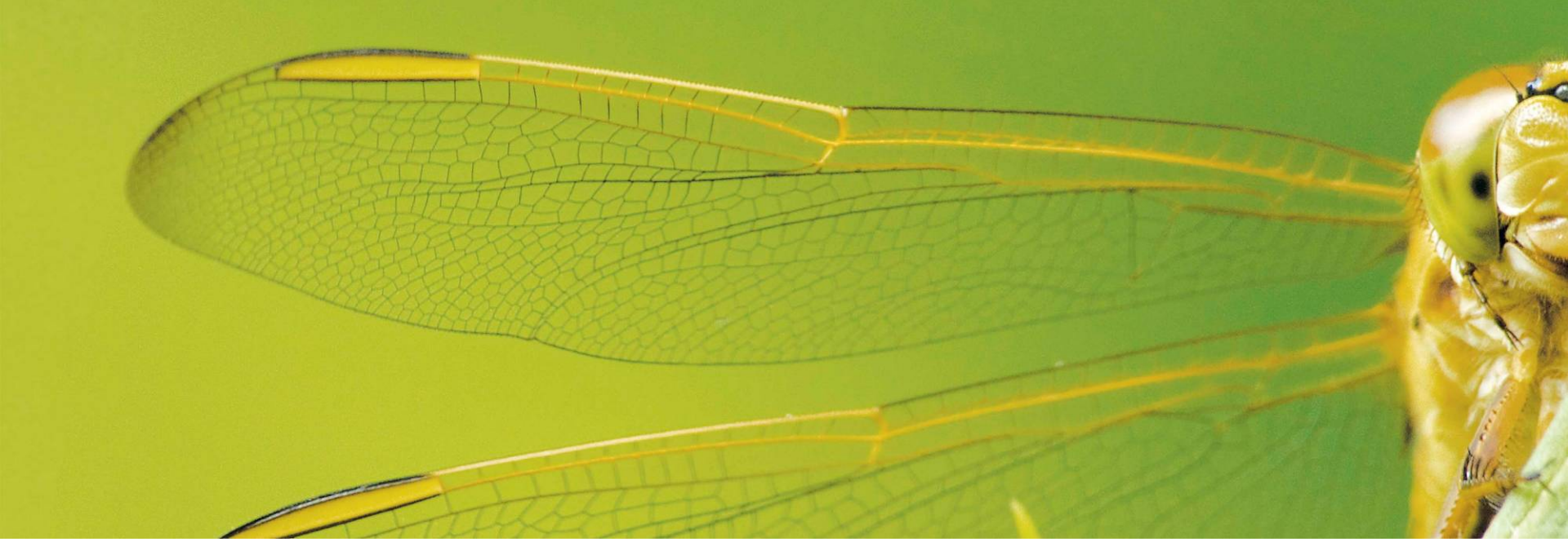


Asterisk

The Open Source Telephony Project

Vortrag von Roman Schneuwly

leave it up to us

albis technologies

Albis Technologies AG
Wer sind wir?

Wir entwickeln für unsere Kunden Produkte

- Von der Idee bis zum marktreifen Produkt

A-Z

- Wir haben 20 Jahre Erfahrung

20 Jahre

- 100 top qualifizierte Ingenieure für:
  Hard-, Software, Systemengineering,
  Hochfrequenz, PCB Design, EMC

+100

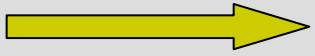
- Management buy out aus der Siemens Schweiz

albis Siemens

What is Asterisk?

albis technologies

- A conversion gateway for . . .
  - physical media (C-T1, PRI, FXO, FSX, IP)
  - protocol (TDM,SIP,H.323,IAX,MGCP,SCCP)
  - codec (G.729,G.711,GSM,ILBC,G.726, etc.)
- Open-source (GPL + exceptions)
- Easily extended with Perl/C/Python/ etc. or apps written (typically C)
- Flexible enough to do almost any telecommunications task (conferencing, recording)
- Runs on a standard platforms (x86 …) (Linux based)



Why Asterisk?

albis technologies

- It's FREE
  - How much are you paying your PBX vendor now?
- Runs on commodity PC hardware
- Broad support for VoIP protocols and hardware
- Easy to interconnect with other boxes
  - Form your own VoIP network
- Configurable to do (almost) whatever you want
  - Tweak it to your needs
  - Write your own code
  - It will still not do your dishes, unfortunately

Session Initiation Protocol (SIP)

albis technologies

- Signalling protocol only
  - Actual media transport handled by RTP
- Protocol developed by IETF, not ITU-T
  - Uses URLs instead of telephone numbers
  - sip:info@albistechnologies.com
- Intended to be a peer-to-peer protocol
- Fairly ubiquitous
  - Most VoIP phones, terminal adapters, etc speak SIP
- Does not play well with NAT

H.323

albis technologies

- Developed in 1996 by ITU-T
- Far more similar to traditional telephony signaling protocols than SIP
- Uses RTP for media transport
- Used internally by interexchange carriers
- Fairly unpopular in the do-it-yourself VoIP world
    - Difficult to implement in software
    - Major pain in the ass to get working correctly

Inter-Asterisk EXchange (IAX)

albis technologies

- Developed by Mark Spencer of Digium
- Covers both signaling and media transport
    - Streamlined, simple protocol
- Does not suffer from NAT traversal issues
- Data and signaling happen via UDP on port 4569
- Well-supported by Asterisk
- Support in terminal equipment is rare
    - Digium IAXy terminal adapter speaks IAX
- Preferred protocol for many PSTN termination providers

Codecs supported by Asterisk

albis technologies

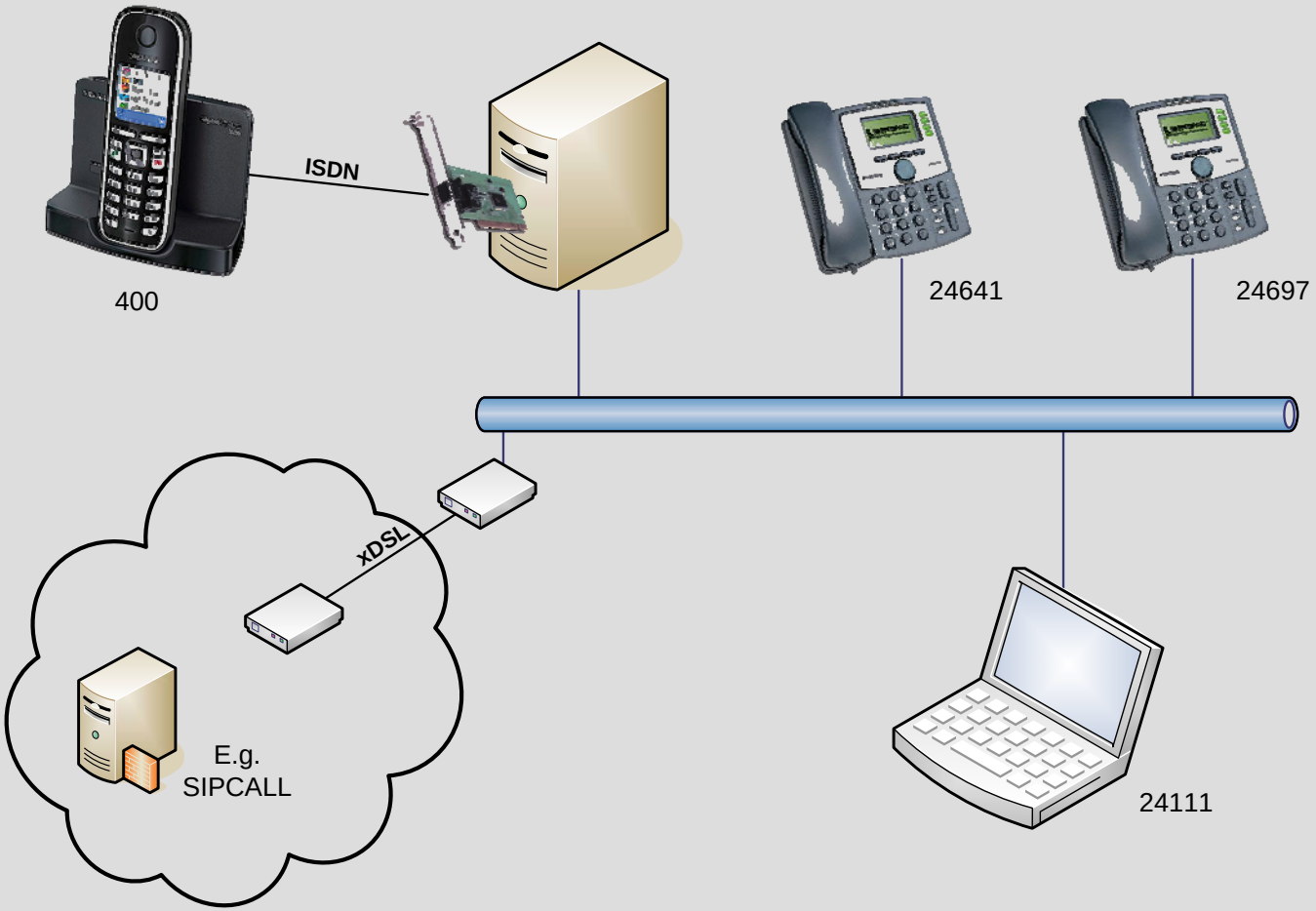
- G.711
  - 64kbps µ-law or a-law companding
- G.726
  - 32kbps Adaptive Differential Pulse Code Modulation
- G.729
  - 8kbps Conjugate-Structure Algebraic Code-Excited
- Linear Prediction
  - Requires a license
- GSM
  - 13kbps Regular Pulse Excitation Long-Term Prediction

Codecs supported by Asterisk

albis technologies

- Internet Low Bandwidth Codec (iLBC)
  - 13.3kbps Linear Predictive Coding
  - This is the codec used by Skype
- Speex
  - 13.3kbps Code-Excited Linear Prediction
  - Open Source codec
- LPC10
  - 2.4kbps Linear Predictive Coding

Asterisk Network



SIP Client Settings

- Set up SIP clients configuration:
  - Identification (username)
  - Secret (password)
  - Network Type …

\etc\asterisk\sip.conf

```
[testuser1]
secret=111269
callerid=Mueller Peter<24641>
type=friend
host=dynamic

[testuser2]
secret=120119
callerid=Sieber Hans <24697>
type=friend
host=dynamic
```

Dialplan

albis technologies

- Set up Asterisk Dialplan
  - Phone Nr 24697 → Forwarded to SIP Client with ID testuser2
  - Phone Nr 24641 → Forwarded to SIP Client with ID testuser1
  - Phone Nr 1001 → Automatically Plays something
  - Phone Nr 400 → Forwarded to ISDN MSN 400

\etc\asterisk\extensions.conf

```
[default]

exten => 1001,1,Answer()
exten => 1001,2,Playback(hello-world)
exten => 1001,3,Hangup()

exten => 24697,1,Dial(SIP/testuser2)
exten => 24641,1,Dial(SIP/testuser1)

exten => 400,1,Dial(misdn/1/400,10,Ttr)
```

External Call Settings
SIP Provider

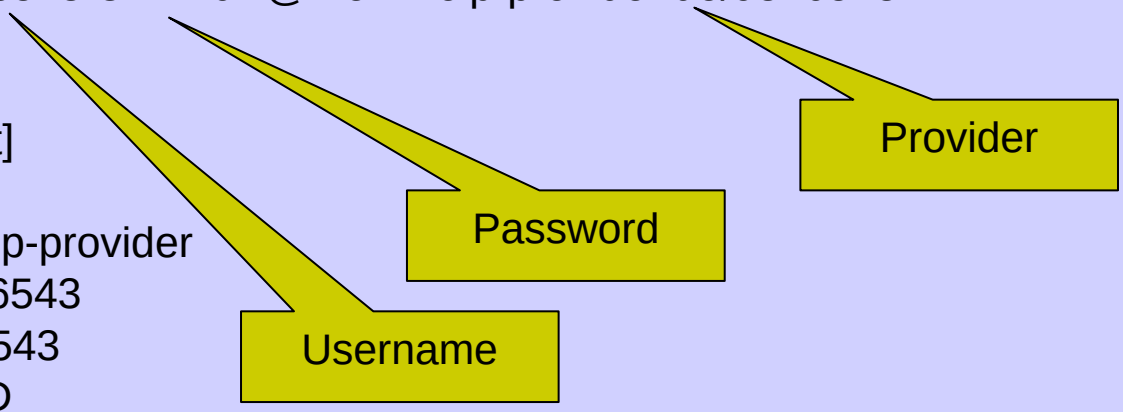
albis technologies

- Set up Account Settings for SIP Provider

\etc\asterisk\sip.conf

```
register => 9876543:UHDZJD@mein-voip-provider.de/9876543


[ext-sip-account]
type=friend
context=von-voip-provider
username=9876543
fromuser=9876543
secret=UHDZJD
host=mein-voip-provider.de
fromdomain=mein-voip-provider.de
qualify=yes
insecure=port,invite
nat=yes
```



External Call Settings
Calling Rules

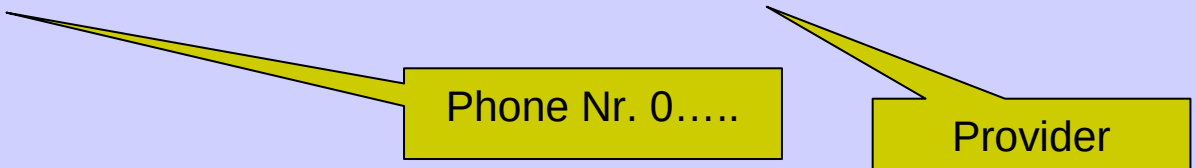
- Outgoing Calls
  - All Numbers with 0…. are forwarded to ext-sip-account
- Incoming Calls
  - Incoming calls are forwarded to local SIP Client (ID=testuser2)

\etc\asterisk\extensions.conf

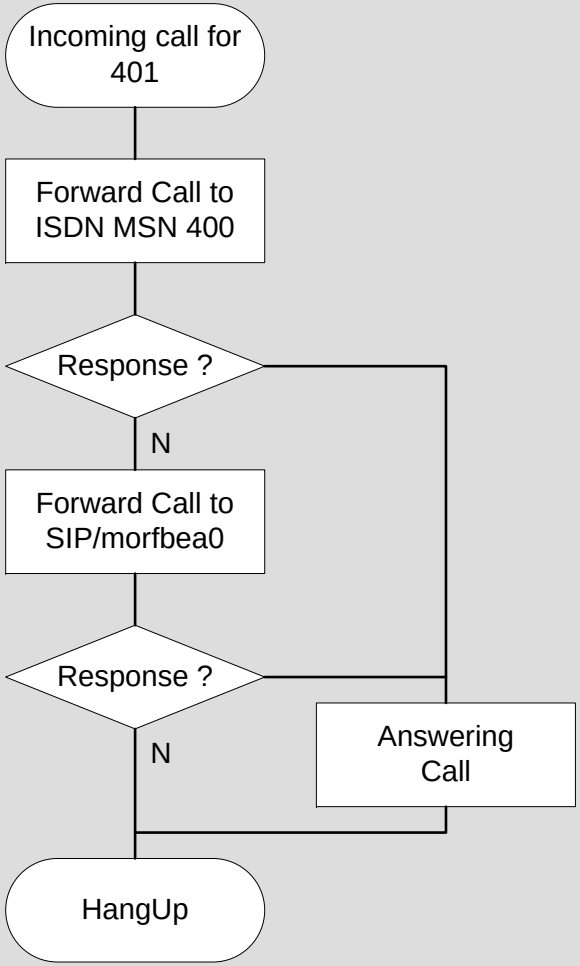
```
exten => _0[1-9].,1,Dial(SIP/${EXTEN}@ext-sip-account)

[von-voip-provider]

exten => 9876543,1,Dial(SIP/testuser2)
```



Call chain …

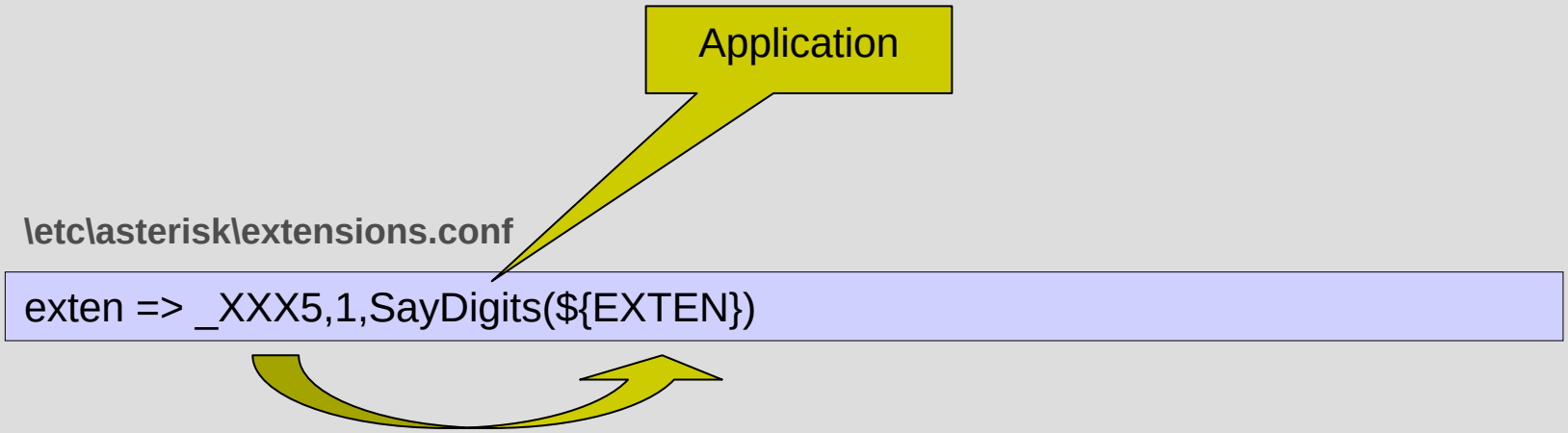



```
exten => 401,1,Dial(misdn/1/400,10,Ttr)
exten => 401,n,Dial(SIP/testuser2)
exten => 401,n,Hangup
```

Asterisk Applications I

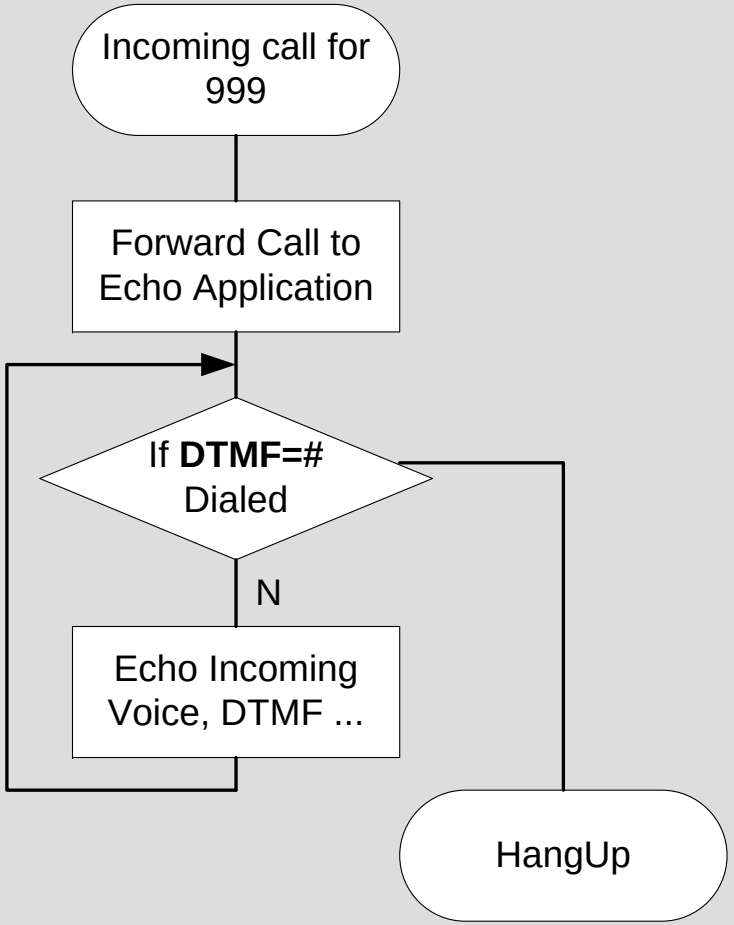
albis technologies

- Interface for adding functionality to Asterisk
- Cross-Language
  - Perl
  - C
  - PHP ...
- Example SayDigits …



Asterisk Applications II
Echo

- Echo Example …



\etc\asterisk\extensions.conf

```
exten => 999,1,Answer()
exten => 999,n,Echo()
exten => 999,n,Hangup()
```

Echo Application (Implementation)

albis technologies

```
while (ast_waitfor(chan, -1) > -1)
{
        struct ast_frame *f = ast_read(chan);
        if (!f)
        {
                break;
        }
        f->delivery.tv_sec = 0;
        f->delivery.tv_usec = 0;
        if (ast_write(chan, f))
        {
                ast_frfree(f);
                goto end;
        }
        if ((f->frametype == AST_FRAME_DTMF) && (f->subclass == '#'))
        {
                res = 0;
                ast_frfree(f);
                goto end;
        }
        ast_frfree(f);
}
```

Advantages of Asterisk

albis technologies

- Open Source Software (Free)
- Supports different protocols:
  - IAX
  - H.323
  - Session Initiation Protocol (SIP),
- Interoperates with different equipments:
  - Standard SIP phones (Cisco)
  - FXS and FXO devices (PSTN)
  - PRI (Primary Rate Interfaces) 2mbps
- Modularity / Features
  - Voicemail … and lots more
  - Own applications can be implemented

Disadvantages of Asterisk

albis technologies

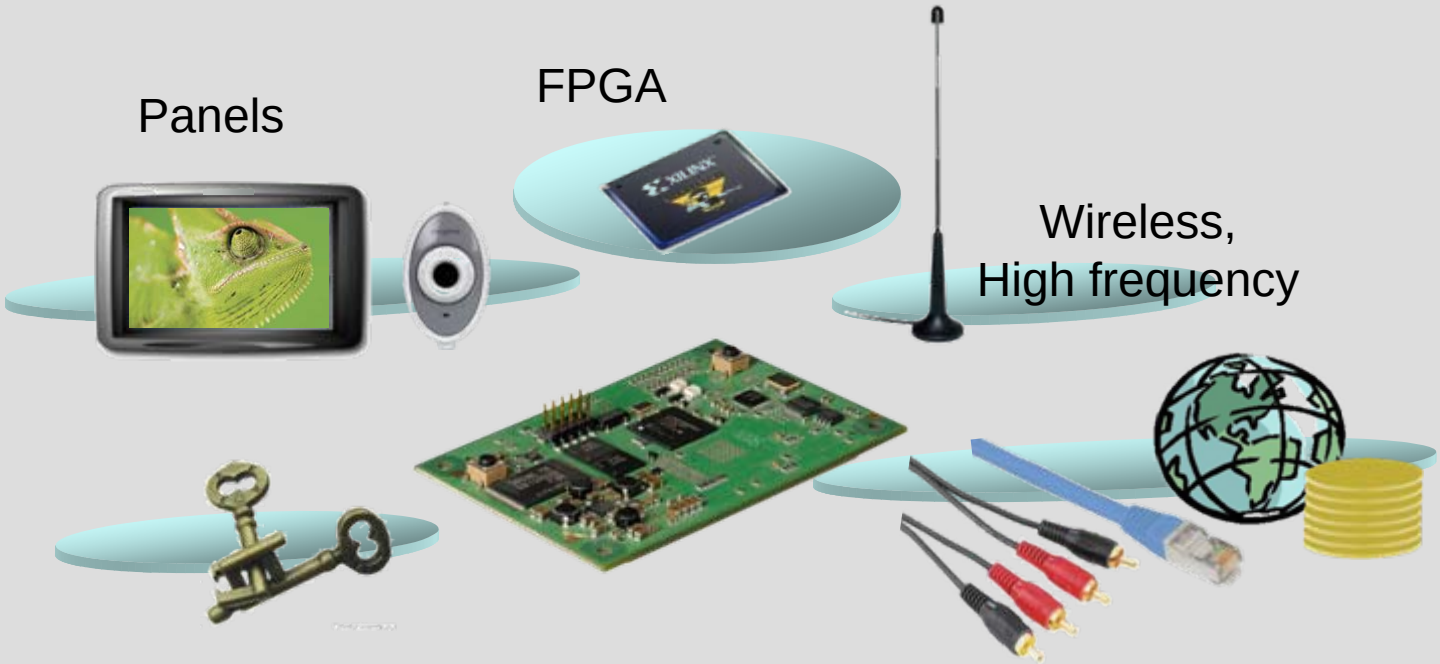
- Extra complexity
- TCO has to be taken into account
  - Installation is „cheap“
  - Support costs could be quite high
- NAT and SIP is no good choice

References

- Asterisk – The Future of Telephony
- http://www.das-asterisk-buch.de/
- DC_13-Carlson-Ratchet.pdf
- astercon-intro-to-asterisk.pdf
- http://de.wikipedia.org/wiki/H.323

R&D: Center of Competences

Embedded Systems



Sie bestimmen, was Sie aus unserem Baukasten für Ihr embedded System benötigen

Leave it up to us

- **Embedded Systems (HW &SW)**
- **Communication**
- **Wireless, Hoch Frequenz**
- **Embedded Security**
- **Akkreditierte Certifizierung für EMC & safety**

Kontakt Albis Technologies AG
Marco Tölle
8047 Zürich
Phone +41 58 252 4777

development@albistechnologies.com
www.albistechnologies.com/services/research