VOIP with Asterisk & Perl

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The Elements of VOIP

- PSTN - “Public Switched Telephone Network”, the pre-Internet phone system: land-lines & cell-phones.

- DID - “Direct Inward Dial”, a *phone number* on the PSTN.

- SIP - “Session Initiation Protocol”, the *ringing, answered and hangup* signals of VOIP.

- RTP - “Real-time Transport Protocol”, the audio (or video) of the VOIP call; also know as: the *media stream*.

- UDP - “User Datagram Protocol”, used for most VOIP packets.
Why UDP?

- Most VOIP systems use the UDP protocol to initiate calls and transport the media stream.

- UDP is used for VOIP packets instead of TCP because:
  - TCP packets might be resent if they are not delivered immediately.
  - TCP packets might be re-ordered, or delayed before reaching the application.

- Most VOIP audio codecs are capable of withstanding some packet loss.
Which audio codec?

Audio codecs encode, and optionally compress, audio signals into binary streams that can be sent over the network.

There are a few commonly used codecs:

- **G.711 (u-law)** - uncompressed, widely supported by carriers.
- **G.729** - compressed, requires a license to use, though widely supported.
- **Speex** - an open source codec for speech, low bandwidth.
- **G.722** - wide-band, high bandwidth, HD codec, supported by only by VOIP phones.
So how does it all work?

**Receive Calls**
PSTN -> VOIP:
Origination

**Place Calls**
VOIP -> PSTN:
Termination

Flowroute provides a gateway to the PSTN, charges apply.

**Flowroute connects calls to / from carriers.**

**The PSTN:**
- SIP Media Server
  - SIP/RTP calls
  - Perl
- Soft-phone
  - SIP/RTP calls
- SIP Phone
  - SIP/RTP calls
- Flowroute
  - SIP/RTP calls
Termination: Placing a call to the PSTN using SIP

Switchboard operators ain’t what they used to be...

Then:

![Switchboard operators](image1.png)

Now:

![Terminator](image2.png)
Origination: Receiving a call from the PSTN via a DID

“Hello”
Network Address Translation:

VOIP & NAT can cause problems

* External devices cannot send packets directly to devices behind a NAT firewall.

* For VOIP there is a grab-bag of tricks that are used to overcome this limitation.

* Asterisk is fairly adept at dealing with the problem.

  * Most modern, consumer-grade routers will work properly with VOIP by default.

  * If problems are encountered:

    * Read documentation at http://voip-info.org/.
Asterisk: an Open Source Media Server

- Asterisk is a daemon that you run on your system to provide SIP and RTP media streaming for VOIP calls.
- Asterisk is a Virtual PBX, which means it is configured by default to be a corporate-style, branch phone system where each phone has an extension: 100, 101, 102, etc...
- Asterisk provides the features to receive inbound SIP calls and route them to either a VOIP phone or a PSTN gateway. Asterisk can also initiate calls programmatically.
- Asterisk can answer calls internally to play sounds, record messages, and create Interactive Voice Response systems (IVRs), etc...
SIP profile to connect to the PSTN through a VOIP service provider (Flowroute):

```
[flowroute]
host=sip.flowroute.com
username=47346967
secret=w9cslk38w12
context=inbound
type=friend
qualify=yes
dtmfmode=rfc2833
nat=no
insecure=port,invite
canreinvite=no
allow=ulaw
allow=g729
```

- The name of the SIP profile
- SIP host to connect to
- SIP Login
- SIP Password
- Everybody is a “friend” of Asterisk these days
- Flowroute is not behind a NAT
- Periodically check if Flowroute is reachable
- Which type of dialpad key-press signals to use
- Dialplan context to use by default for this profile
- Able to redirect RTP media stream to a different host
- Limit media codecs
- Allow uncompressed 8Hz U-law codec
- Allow compressed g729 codec
- Accept calls from any registered IP
Asterisk Config: /etc/asterisk/sip.conf

+ SIP profile for a VOIP desk phone:

```plaintext
[100]
host=dynamic            ; Phone will get its IP address using DHCP
secret=c83jx73jx
type=friend
nat=yes                 ; The phone might be behind a NAT
canreinvite=no
disallow=all
allow=ulaw
allow=g729
context=localexts       ; Default context “localexts” for internal phones
insecure=port,invite
qualify=yes
```
Context for outbound calls from internal phones:

```plaintext
context localexts {
    _NXXXXXXXXX => {
        Set(CALLERID(num)=13104561234);
        Dial(SIP/1${EXTEN}@flowroute);
    };
    100 => {
        Dial(SIP/100,45);
        Voicemail(100@default);
    };
    101 => {
        Dial(SIP/100,45);
        Voicemail(100@default);
    };
    102 => {
        Dial(SIP/100,45);
        Voicemail(100@default);
    };
    301 => {
        Agi(agi://127.0.0.1:7788/fastcgi);
    };
};
```
Asterisk Config: /etc/asterisk/extensions.ael (More Dialplan)

+ Contexts for inbound calls from the PSTN and an IVR prompt for the company:

```plaintext
context inbound {
    13104561234 => {
        Answer();
        goto voiceprompt,s,begin;
    },
};
context voiceprompt {
begin:
    s => {
        Wait(1);
        Background(main-company-prompt); // Play the message
        WaitExten(15); // Wait for an extension, otherwise
        Dial(SIP/100&SIP/101/&SIP/102,25); // Ring all these phones at once
        Voicemail(100);
    },
    _1XX => {
        Dial(SIP/${EXTEN},25);
        Voicemail(${EXTEN});
    },
};
```

Wednesday, December 7, 2011
Allow your programs to access the Asterisk Manager Event API:

```
[astmgr]
secret = v2ovm39clg8
deny=0.0.0.0/0.0.0.0
permit=127.0.0.1/255.255.255.255
read = system,call,log,verbose,agent,user,config,dtmf,reporting,cdr,dialplan
write = system,call,agent,user,config,command,reporting,originate
```

Middle Manager Bob
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Perl & Asterisk: 2 Interfaces

- AMI - Asterisk Manager Interface
  - The “Manager” interface allows external programs to monitor call-related events inside Asterisk. Perl connects to an event socket and listens for events.
  - The AMI interface can also be used to initiate events, like originating a call, transferring a call, and more. Perl will connect a socket, issue a command, then disconnects.

- AGI - Asterisk Gateway Interface
  - The AGI interface is used to control calls from inside the dialplan. The “Agi(...)” application will connect to a Perl daemon. Once connected to the daemon, the call progress will be controlled by Perl.
Perl Modules:

**AMI Modules:**

* Asterisk::AMI - Newest module, OO-interface, recommended.
* Asterisk::Manager - Reference module, still works.

**AGI Modules:**

* Asterisk::FastAGI - Based on Net::Server, recommended.
* Asterisk::AGI - Reference module, same functions as Asterisk::FastAGI.
Show me some code!
#!/usr/bin/perl
use Asterisk::AGI;
use Net::Ping::External qw(ping);
$AGI = new Asterisk::AGI;
my $input = $AGI->ReadParse();
my $finished = 0;
$AGI->exec('Festival', '"Enter the eye-p address you wish to ping."');
my $ipaddr = ''; my $x = 0;
while (!$finished) {
    my $input = chr($AGI->wait_for_digit('5000'));
    if ($input =~ /[0-9\*\#]/) {
        # pressed something
        if ($input =~ /[\*\#]/) {
            # pressed * or #
            $x++;
            if ($x > 3) {
                $finished = 1;
            } else {
                $ipaddr .= '.';
            }
        } else {
            # pressed a digit
            $ipaddr .= $input;
        }
    } else {
        # must have timed out
        $finished = 1;
    }
    if (length($ipaddr) > 14) {
        $finished = 1;
    }
}
if ($ipaddr !~ \d{1,3}\.\d{1,3}\.\d{1,3}\.\d{1,3}/) {
    $AGI->exec('Festival', "Invalid Address: $ipaddr");
    exit 0;
}
$AGI->exec('Festival', "Please wait");
if (ping(host => "$ipaddr", timeout => 2)) {
    $AGI->exec('Festival', "Host is up");
} else {
    $AGI->exec('Festival', "Host is down");
}

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