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.

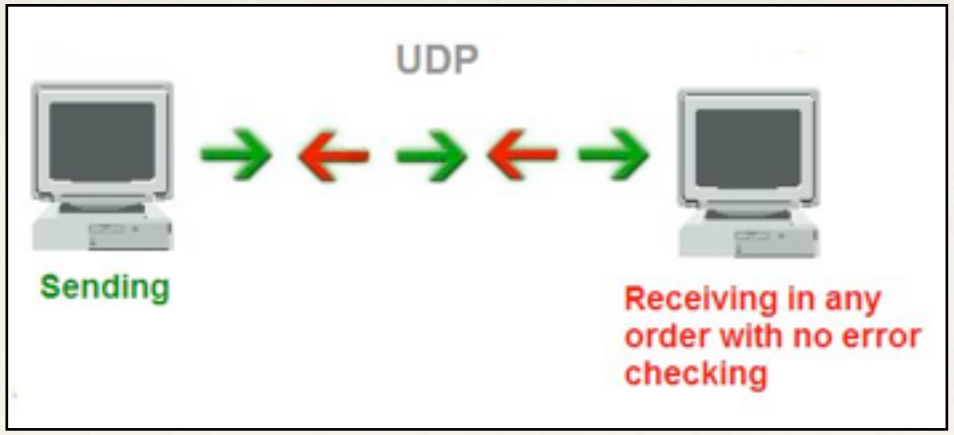
- DID - “Direct Inward Dial”, a *phone number* on the PSTN.

- 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.

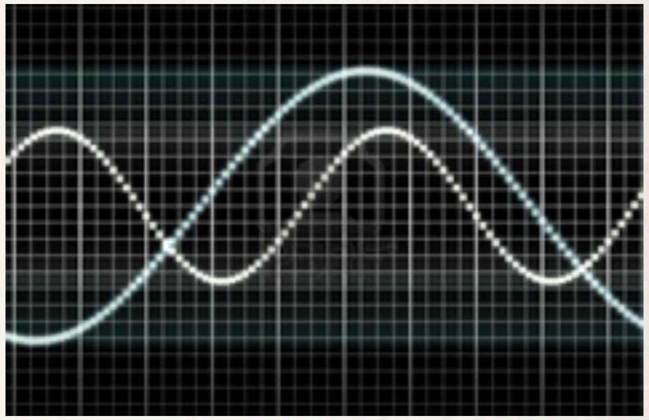
Why UDP?



- 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.

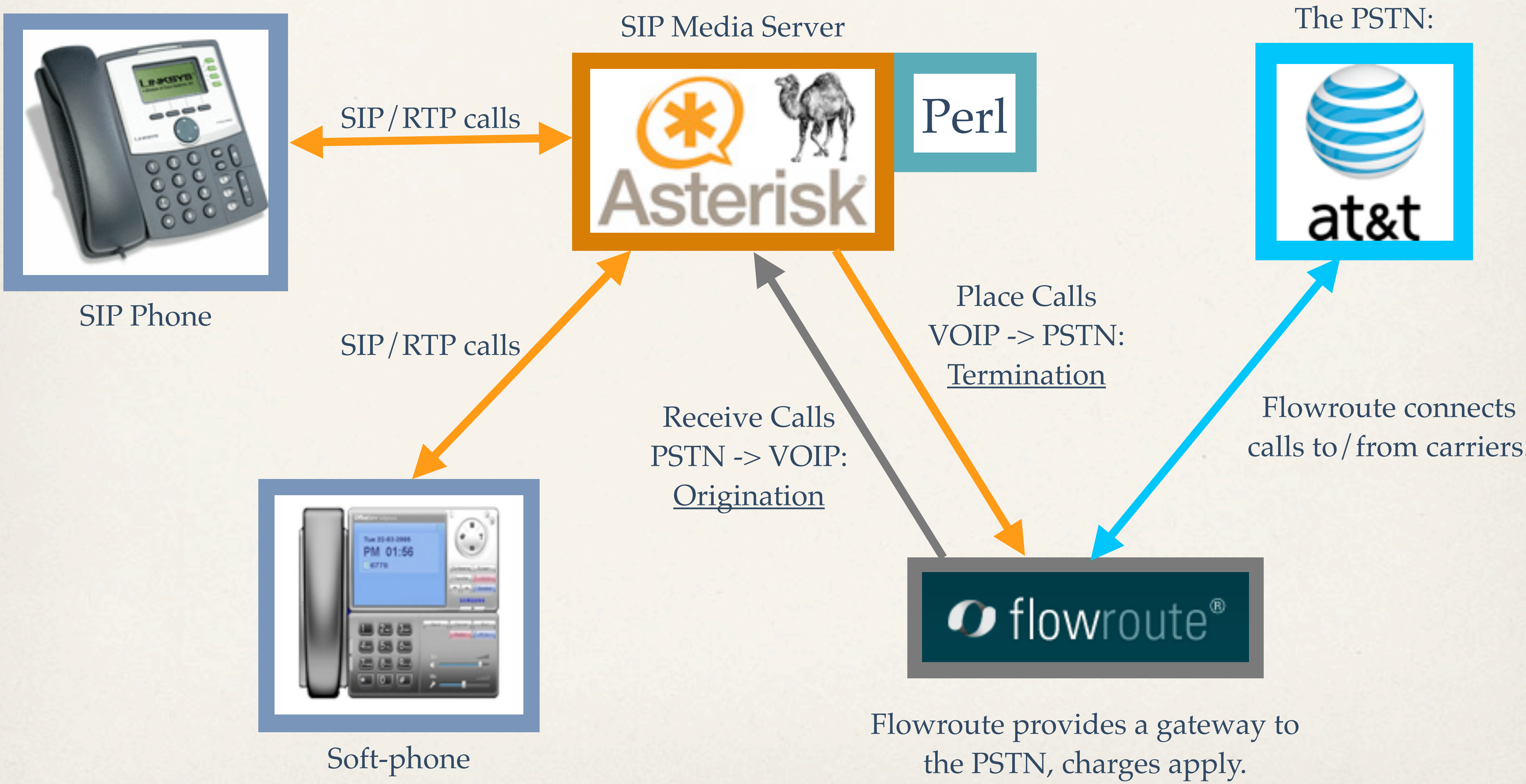
Which audio codec?

- 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?

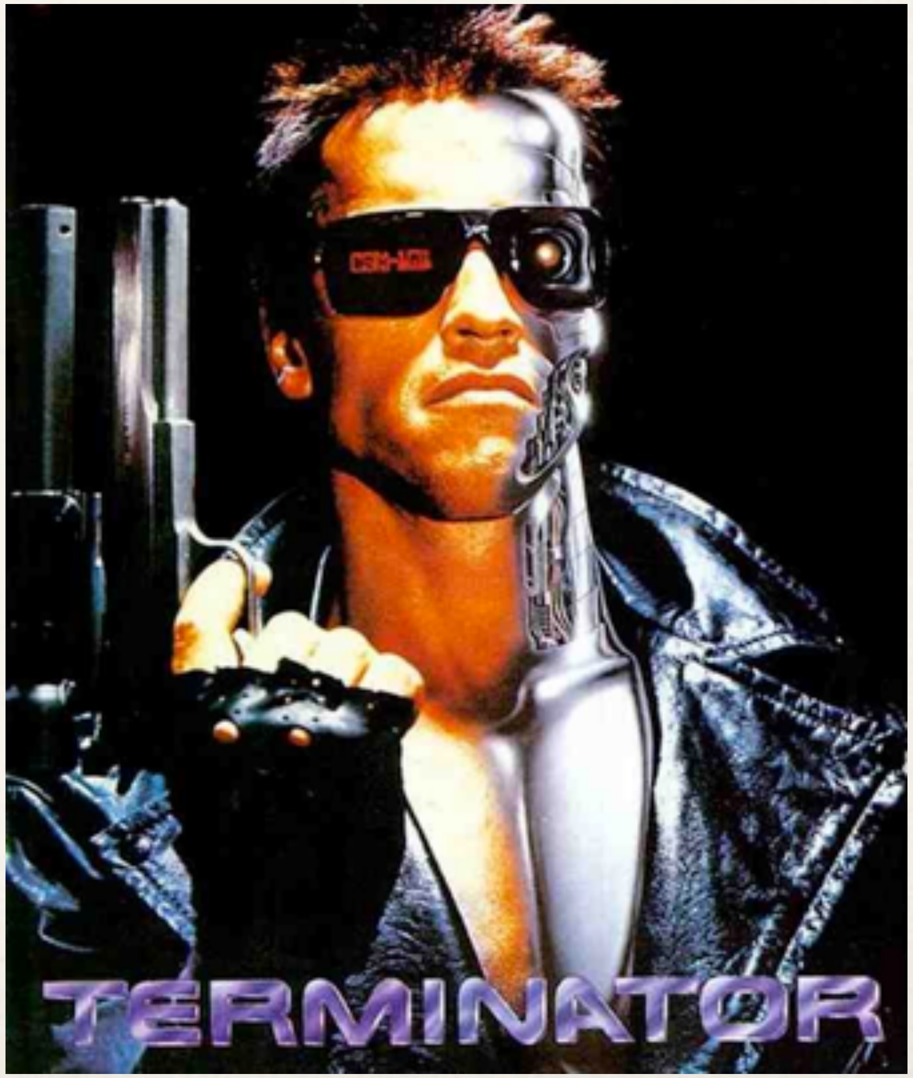


Termination: Placing a call to the PSTN using SIP

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

Then:

Now:

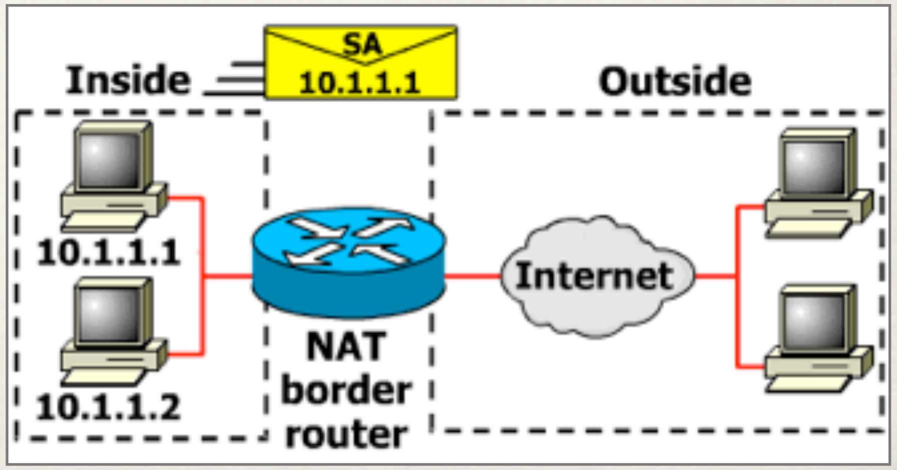


Origination: Receiving a call from the PSTN via a DID

"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...

Asterisk Config: /etc/asterisk/sip.conf

- SIP profile to connect to the PSTN through a VOIP service provider (Flowroute):

```
[flowroute]                 ; The name of the SIP profile
host=sip.flowroute.com      ; SIP host to connect to
username=47346967           ; SIP Login
secret=w9cslk38w12          ; SIP Password
type=friend                 ; Everybody is a "friend" of Asterisk these days
nat=no                      ; Flowroute is not behind a NAT
qualify=yes                 ; Periodically check if Flowroute is reachable
dtmfmode=rfc2833            ; Which type of dialpad key-press signals to use
context=inbound             ; Dialplan context to use by default for this profile
canreinvite=no              ; Able to redirect RTP media stream to a different host
disallow=all                ; Limit media codecs
allow=ulaw                  ; Allow uncompressed 8Hz U-law codec
allow=g729                  ; Allow compressed g729 codec
insecure=port,invite        ; Accept calls from any registered IP
```

Asterisk Config: /etc/asterisk/sip.conf

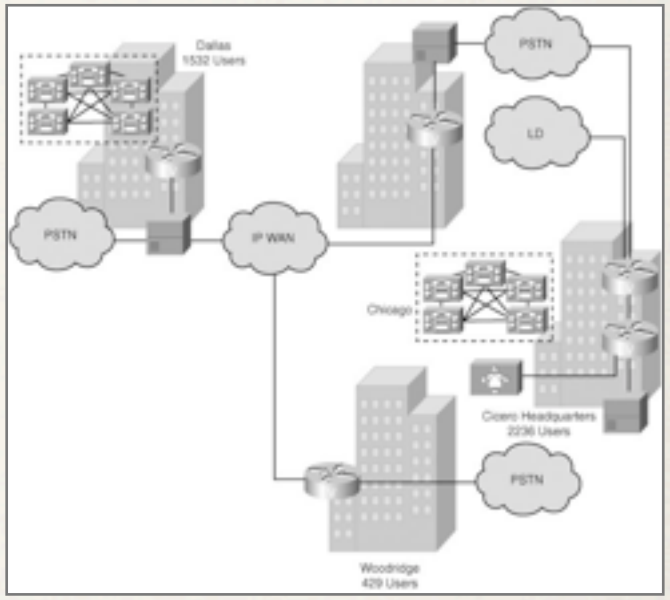
- SIP profile for a VOIP desk phone:

```
[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
```

Asterisk Config: /etc/asterisk/extensions.ael (The "Dialplan")

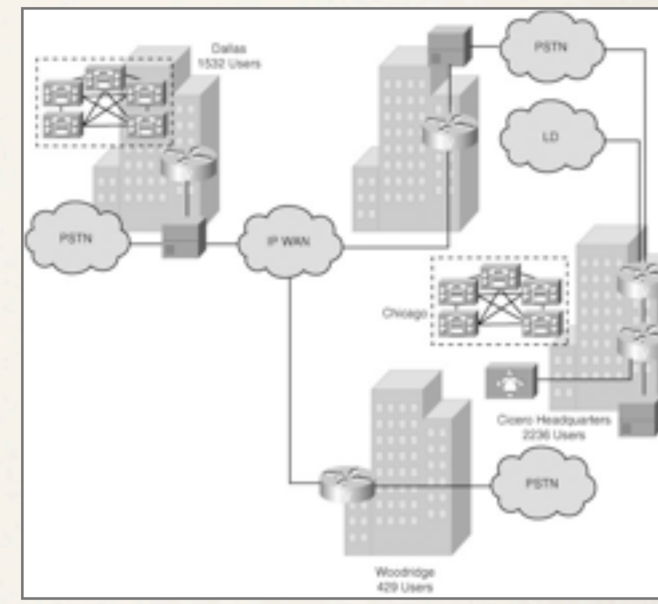
- Context for outbound calls from internal phones:

```
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:




```
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);
    },
};
```

Asterisk Config: /etc/asterisk/manager.conf

- 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

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!

?

```perl
#!/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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