Asterisk update: What's coming down the line in 12?

David Duffett
Working with the Worldwide Asterisk Community

Our agenda for today

- Recap: What is Asterisk?
- Recap: Who are Digium?
- How are the priorities set?
- Asterisk 1.8, 10 and 11
- New stuff, including preliminary support for WebRTC
- WebRTC demo
- Looking into the future - Asterisk 12

Recap: About Asterisk?

- The most mature and established Open Source Communications Engine
- 2+ Million Downloads Each Year
- Over 1 Million Production Deployments
- 86,000+ Registered Community Members
- Under Development Since 1999

Recap: Who are Digium?

digium
The Asterisk Company

- The commercial company behind Asterisk
- Started by Mark Spencer
- Employ lots of software developers to work on code – that we give away!!
- Based in Huntsville, Alabama, USA
- Income comes from Asterisk hardware (including cards, gateways and PHONES), training and subscription support
- Every time you buy DIGIUM you are helping the Asterisk Open Source project.

Roger Bannister

digium
The Asterisk Company

Roger Bannister

- The first man in the world to run 1 mile (or 1.6 km) in less than 4 minutes
- Before he did it (on 6 May 1954), people said it could not be done
- After he did it, many people ran 1 mile in less than 4 minutes (his record lasted 46 days)

Open Source Telephony before 1999

digium
The Asterisk Company

- ??????

Asterisk started in 1999

digium
The Asterisk Company

- Since then…
- In no particular order:
  - Kamailio
  - FreeSwitch
  - CallWeaver
  - Yate
  - sipXecs
  - OpenSIPS
  - …

Phones For Asterisk

- **Digium D40, D50, D70 – Built For Asterisk Systems**
- **Super Simple Provisioning**
  - **Discovery via mDNS / Bonjour**
  - **Uses SIP channel**
- **Integrated Applications:**
  - **Visual Voicemail**
  - **Visual Call Parking**
  - **Queue Management**
  - **Presence / Status**
  - **Call Deflection / Direct VM**

Phones For Asterisk

digium
The Asterisk Company

- **Perhaps more importantly…**
- **A JavaScript API means that you can write applications directly on the phone!!**

What Does Asterisk Do?

digium
The Asterisk Company

- **Plumbing:**
  - Protocol Implementations
  - Media I/O
  - Media Management
  - Session Management
  - Intermediary Services
- **Applications:**
  - Call Routing ("Dialplan")
  - Voicemail
  - Conference Bridge
  - Queues
  - Automated Attendant
  - Directory
  - Call Parking

How Do You Use Asterisk?

digium
The Asterisk Company

- **As An Application Unit Itself:**
  - Hand-Crafted “One-Off” Implementations
  - Dialplan Script Programming + Configuration Files
  - PBX, VoIP Gateway, IVR, ACD, Etc.
- **As A Tool-Kit / Engine:**
  - Use External Interfaces: AGI, AMI
  - “Wrap” Asterisk To Manage
  - Build Custom Applications & Solutions
  - Create Communications Products

Asterisk Releases – 1.8 LTS

digium
The Asterisk Company

- **Asterisk 1.8 LTS**
  - Released 2 Year Ago (AstriCon 2010)
  - Long Term Support (LTS)
  - Added:
    - Secure Calling (SRTP / SIP TLS)
    - IPv6 Support
    - Calendar Integration
    - XMPP Device State
    - Channel Event Logging
    - Google Voice / Chat / Talk Calling (Sort Of)
    - “ISDN” Features (AOC, CCSS, CPID)

Asterisk Releases - Asterisk 10

digium
The Asterisk Company

- **Asterisk 10**
  - **Released ~1 year ago (AstriCon 2011)**
  - **Standard Release (1 Year Lifespan)**
  - **Added:**
    - **HD Media Engine**
    - **New Codecs**
    - **ConfBridge HD Audio / Video conferencing**
    - **T.38 Gateway**
    - Text Message Routing

Asterisk 11

digium
The Asterisk Company

- LTS Release
  - 5 years of full support
  - 1 additional year of security support
  - Takes you to 2018
- LTS = Stability NOT Features
  - Architectural-level bug fixes
  - Performance enhancements
  - Significant refactoring of existing features / functions

New In Asterisk 11

digium
The Asterisk Company

- **Chan_motif – Google / Jingle / XMPP Done Right**
  - Combines chan_google and chan_jingle into a single driver
  - Uses completely refactored XMPP engine (res_xmpp)
  - More stable / less difficult to keep up with Google

- **WebSockets Support For SIP**
  - SIP uses multiple transports: UDP, TCP, TLS now WS
  - New protocol used by web applications for bi-directional, asynchronous communications
  - Integrated into onboard HTTP server in Asterisk

Why WebSockets?

digium
The Asterisk Company

- WebRTC – A New Paradigm For Communications
  - Adds real-time communications to web browsers
    - Audio (Speakers / Microphone)
    - Video (Display / Camera)
  - Implements tools for media session management
    - NAT traversal (STUN, TURN, ICE)
    - Codecs (G.711, Opus)
  - Defines JavaScript APIs for media access, peer connection
  - Leaves the signaling protocol / process open to the application developer

SIP + WebSockets + WebRTC

digium
The Asterisk Company

- **Adding WebSockets to Asterisk enables**
  - **SIP over WebSockets (available in Asterisk 11)**
  - **XMPP / Jingle over WebSockets (future version)**
  - **??? Over WebSockets (as soon as you like)**
- **Other WebRTC changes**
  - **Improved RTP / SRTP handling**
  - **ICE support**
  - **SDP improvements**

This Is Revolutionary Stuff!

- Instantly VoIP enabling every browser in the world
- No software to install
- Interoperability with existing VoIP technologies using RTP
  - SIP
- Open standards: anyone can play
- True unified communications
  - Voice + Video Calling
  - Screen Sharing
  - Conferencing

Next Up: Asterisk 12

digium
The Asterisk Company

- Asterisk is 13 years old, most of the ‘fun’ and ‘sexy’ stuff is done!
- So we are at a point of considering optimisation…
- Asterisk has some architectural issues that need addressing
- Asterisk has a very, very large installed base of users
- How to make improvements without breaking things
- We needed to make some big decisions…

How are the priorities set?

- At AstriDevCon – a meeting of Asterisk developers (usually held the day before AstriCon)

At AstriDevCon 2012…

- Two major areas of development were agreed upon:
  - A new SIP channel driver
  - Major API improvements

Asterisk 12 - Summary

- Asterisk is getting even better ☺
- Work for everyones’ long-term benefit has been started!

At AstriDevCon 2012…

- Why a new SIP channel driver?
  - Asterisk currently has a “homegrown” stack
  - Maintainability issues
  - Difficult to extend with advances in SIP protocol

At AstriDevCon 2012…

digium
The Asterisk Company

- What issues do people have with Asterisk's APIs?
  - AMI
    - No stable identifier to channels (masquerades)
    - Too much information for some applications
    - Synchronous media operations make some use cases difficult

At AstriDevCon 2012…

digium
The Asterisk Company

- What issues do people have with Asterisk's APIs?
  - AGI
    - Not unified well with call control
    - These are indicative of core Asterisk problems, not just problems with the interface protocols

SIP

- New SIP Channel Driver
  - Use a known good SIP stack
  - Highly extensible: be flexible for future improvements
  - Leverage better design for performance
    - Considered SofiaSIP, ReSIProcate, PJSIP

SIP

SIP Stack: pjproject

- Was used on Asterisk SCF successfully
- Use knowledge in building Asterisk SCF for Asterisk

Extensible

- Stand alone modules can be plugged into Asterisk's SIP architecture and provide new SIP-based features
- Example: res_sip_rfc3326 – only implements RFC 3326 (Reason Header field)
- Avoids chan_sip's problem of small changes affecting other functionality

Performance Improvements

- Highly multi-threaded (chan_sip is not)
- Allows deployments to scale back SIP functionality to only what they need

API

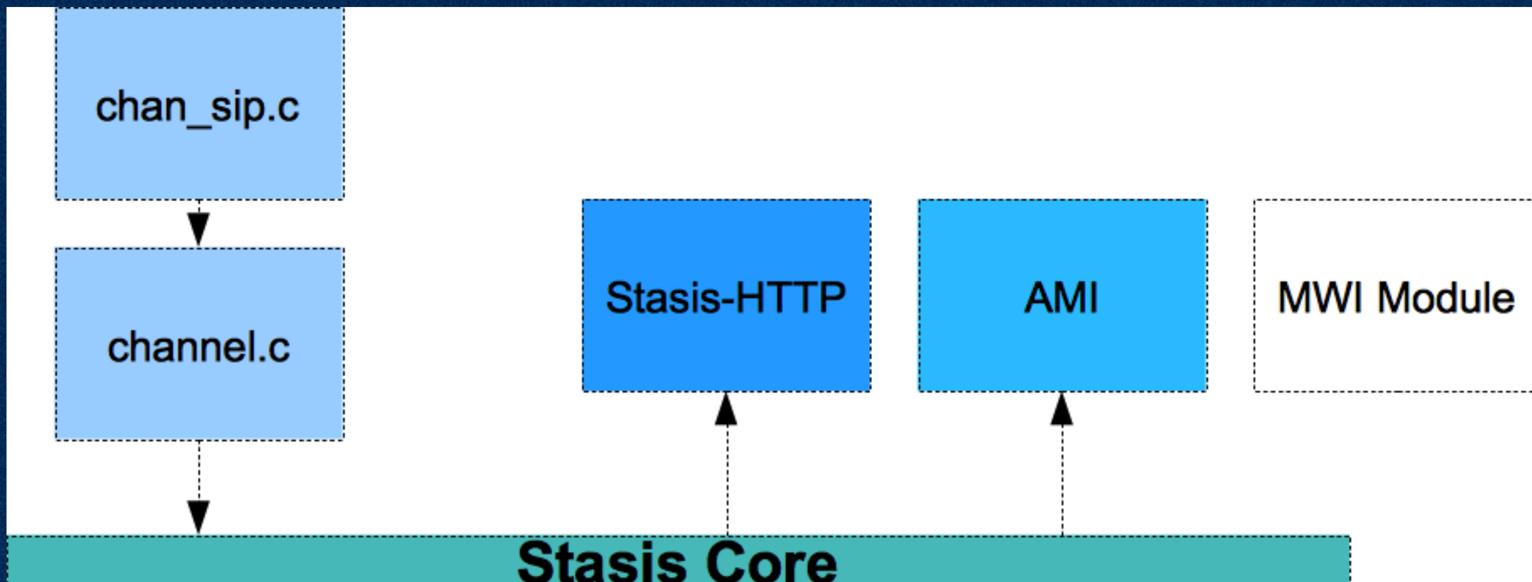
digium
The Asterisk Company

- API Improvements
  - Have a stable handle to channels through the lifetime of communication between the device and Asterisk
  - Do not expose the internal implementation details of Asterisk to third parties
  - Unify call control and channel operations through a friendly API – let users build their own telephony applications easily
  - Fix the core problems in Asterisk that resulted in the interfaces being hard to use

API

- We need a better way to convey information inside Asterisk and to external parties
  - Stasis-Core: the new API in Asterisk that provides a stable way of building on top of information in Asterisk
  - Message bus internal to Asterisk
  - Publish/subscribe to device state, channel state, bridge state, as well as generic messages
  - AMI is re-built on top of Stasis-Core
  - Stasis-HTTP – a new API for building telephony applications – is also build on top of Stasis-Core

API



- A SIP channel is updated and an update is put out onto the Stasis-Core message bus. AMI receives this notification and updates its clients over TCP; Stasis-HTTP receives this and notifies its clients over WebSocket; a module listening for MWI is not notified as this isn't an MWI change

API

- AMI gets a major rework (see Asterisk 12 wiki)
  - Masquerades are no longer 'seen' by third parties, and are reduced significantly in frequency
  - Stable handles to channels that survive for the lifetime of that channel
  - Consistency in information passed in events
  - Publish/subscribe tags that let you only get notifications for things that are associated with items you care about
  - A host of other improvements

API

- Stasis-HTTP: A new API for building Telephony Applications
  - Think of it this way: you don't use Stasis-HTTP to put a channel in an Asterisk application – you use it to build your own applications.

API

digium
The Asterisk Company

- How it works
  - Applications interact with Asterisk over a REST API
  - Events are asynchronous and sent over WebSocket
  - Applications only get events for items in their Stasis-HTTP application
  - Channels in Stasis-HTTP are owned by the application
  - Allows call control as well as channel manipulation
- **Higher level than AMI/AGI, but very powerful**

API

- The implications of the API Improvements on Asterisk is huge
  - Can't get this done without significant work in the core of Asterisk
  - We are hampered by Asterisk's bridging model
  - Masquerades are pervasive – we have to eliminate the need for many of them

API

- Bridging model changes to allow for API sanity
  - Utilize the bridging framework that ConfBridge uses
  - Optimize for two-party bridges, but allow for transitions between two-party bridges and multi-party bridges
  - Simplifies most transfer scenarios such that masquerades aren't necessary

API

- Bridging Changes have a ripple effect
  - CDRs change. Asterisk currently has lots of 'stuff' put in to try and make CDRs work with its current threading/bridge model. It has never worked perfectly due to fundamental design problems. This code no longer exists when we move to the Bridging API.
  - CELs change. Not as much as CDRs, but the lack of masquerades mean some events will occur differently.
  - Queue Logs change. Transfers work differently, bridging works differently.
  - Lots of internal changes

digium
The Asterisk Company

- Specifications for all of the above are being written

API

- This is an opportunity to fix long standing problems in Asterisk
  - But, there is a cost to fixing these problems
  - Asterisk 12 is not like other releases
  - Asterisk 11 was an incremental improvement on Asterisk 10. Asterisk 10's changes – which included major media changes – were mostly transparent. <u>Asterisk 12 will not be transparent</u>.

API

- We will focus on backwards compatibility where possible, but we can't guarantee it
- But we will document what changed and how things should behave

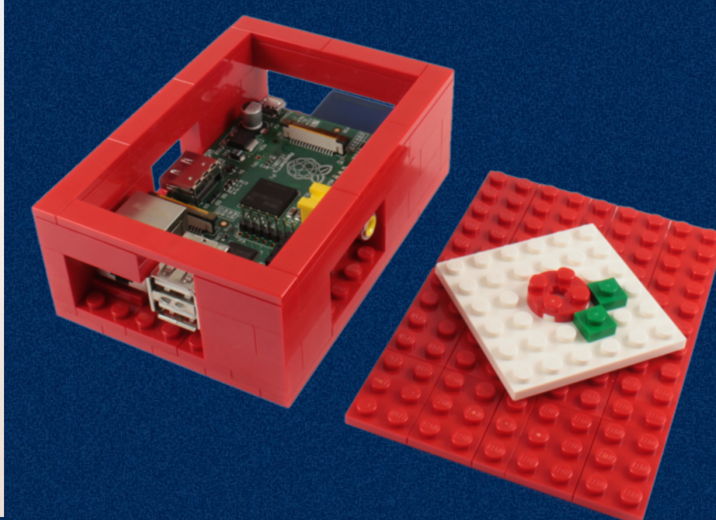
Asterisk 12 - Summary

- Asterisk is getting even better ☺
- Work for everyones' long-term benefit has been started!

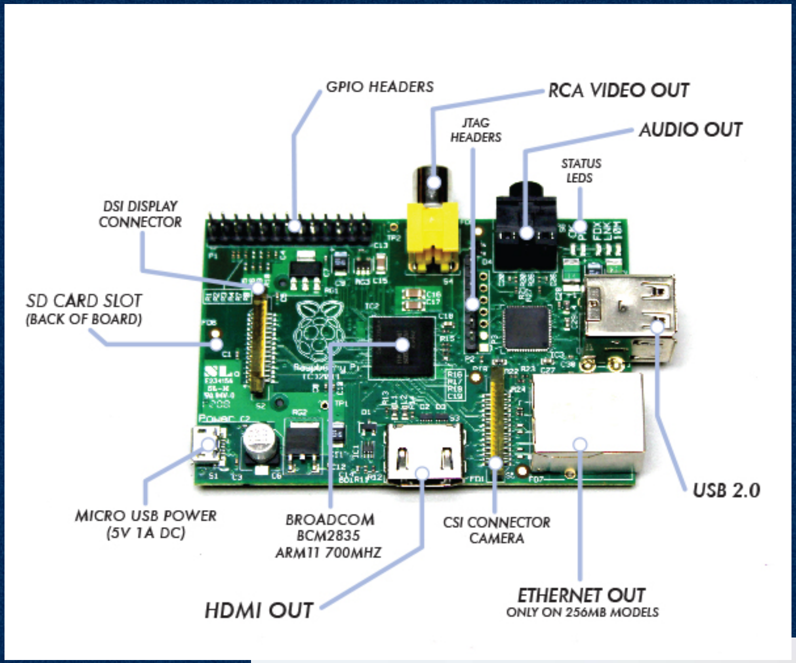
Now for a WebRTC demo on the the Raspberry Pi !!

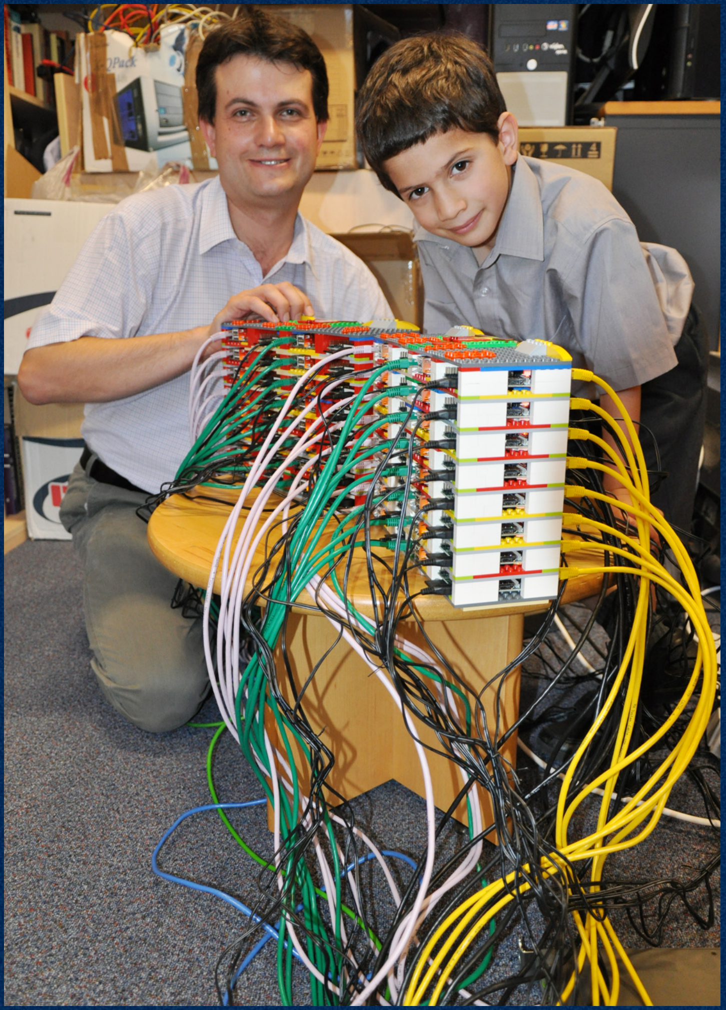
digium
The Asterisk Company

- **What is the Raspberry Pi?**
  - **A small but fully functional ARM based computer that runs Linux**
  - **Originally developed to allow young people to start programming at a very low cost ($40 US)**
  - **A number of Asterisk implementations, including PBX in a Flash/ Incredible Pi by Ward Mundy**



More about the Raspberry Pi





This is my Pi

Rasberry Pi – Model B
512 Mb RAM

Asterisk 11.0.1
DPMA
(Digium Phone Module for Asterisk)
DHCP server

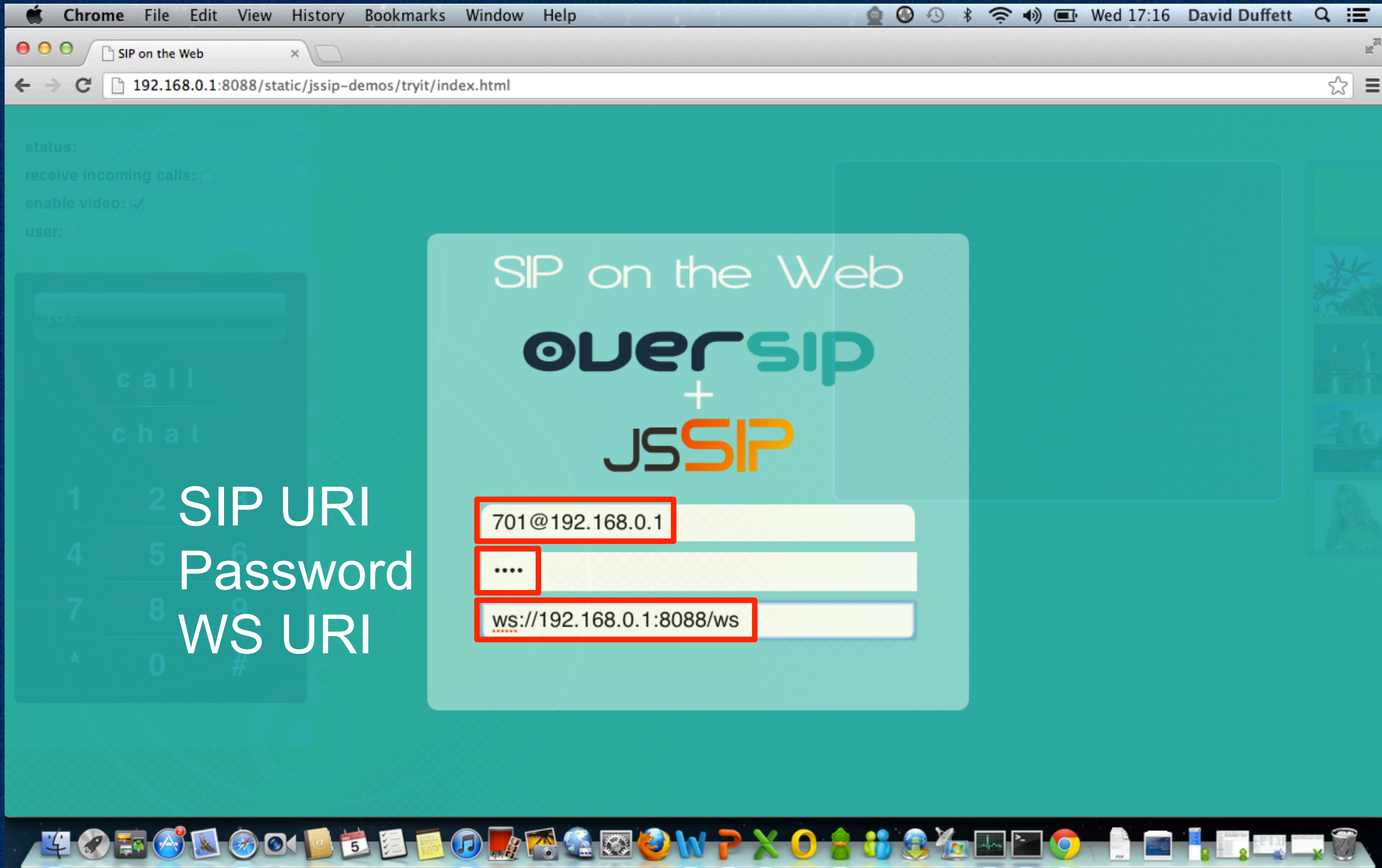
Asterisk is running an HTTP server, Serving JsSIP

digium
The Asterisk Company

- Connect to WiFi network ‘DigiumDemo’
- Browse to http://192.168.0.1:8088/static/jssip-demos/tryit/index.html
- Enter credentials into the 3 fields:
  - SIP URI = 701, 702, 703, 704, 705@192.168.0.1
  - Password = demo
  - WS URI = ws://192.168.0.1:8088/ws

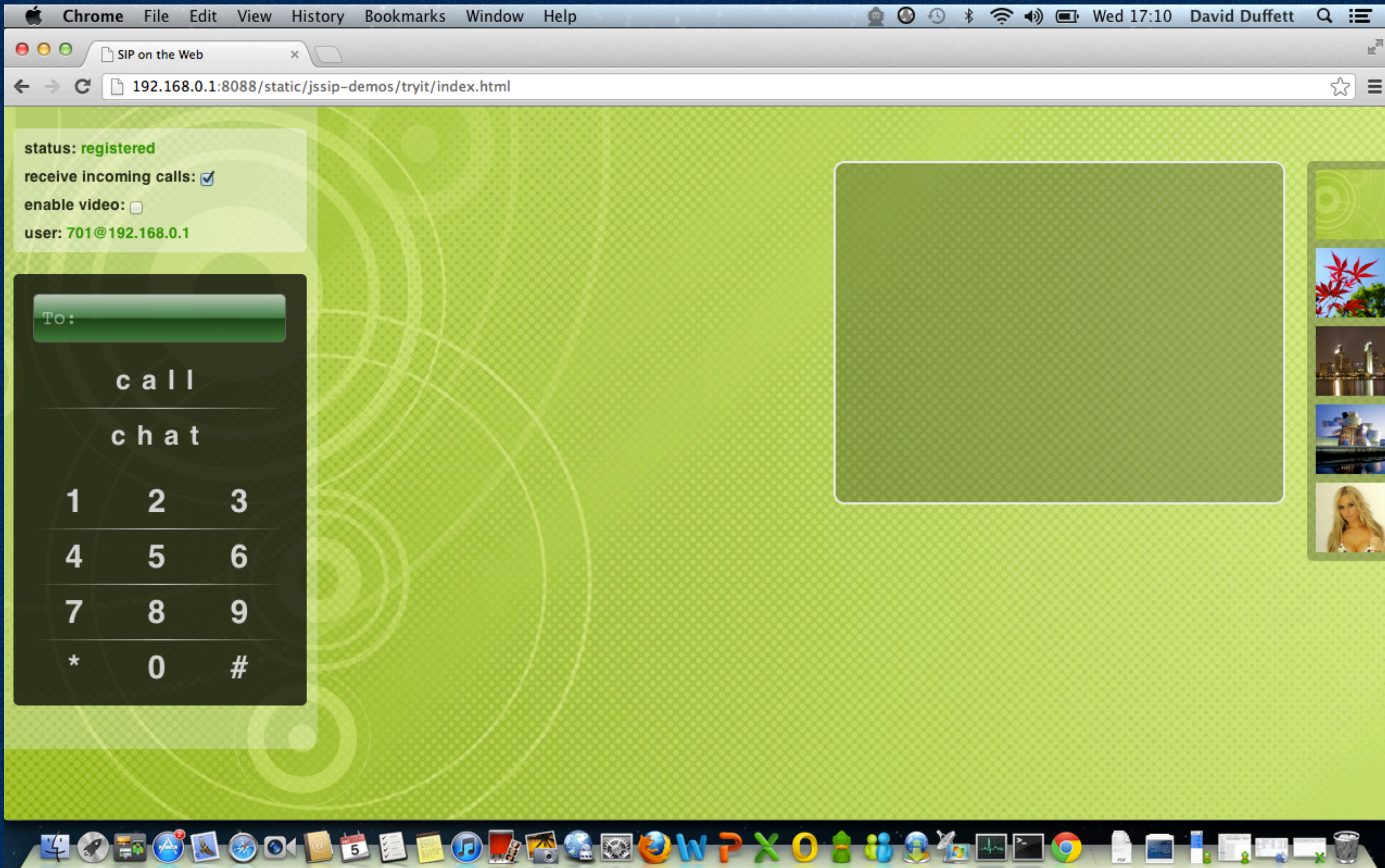
Enter details into this screen

digium
The Asterisk Company

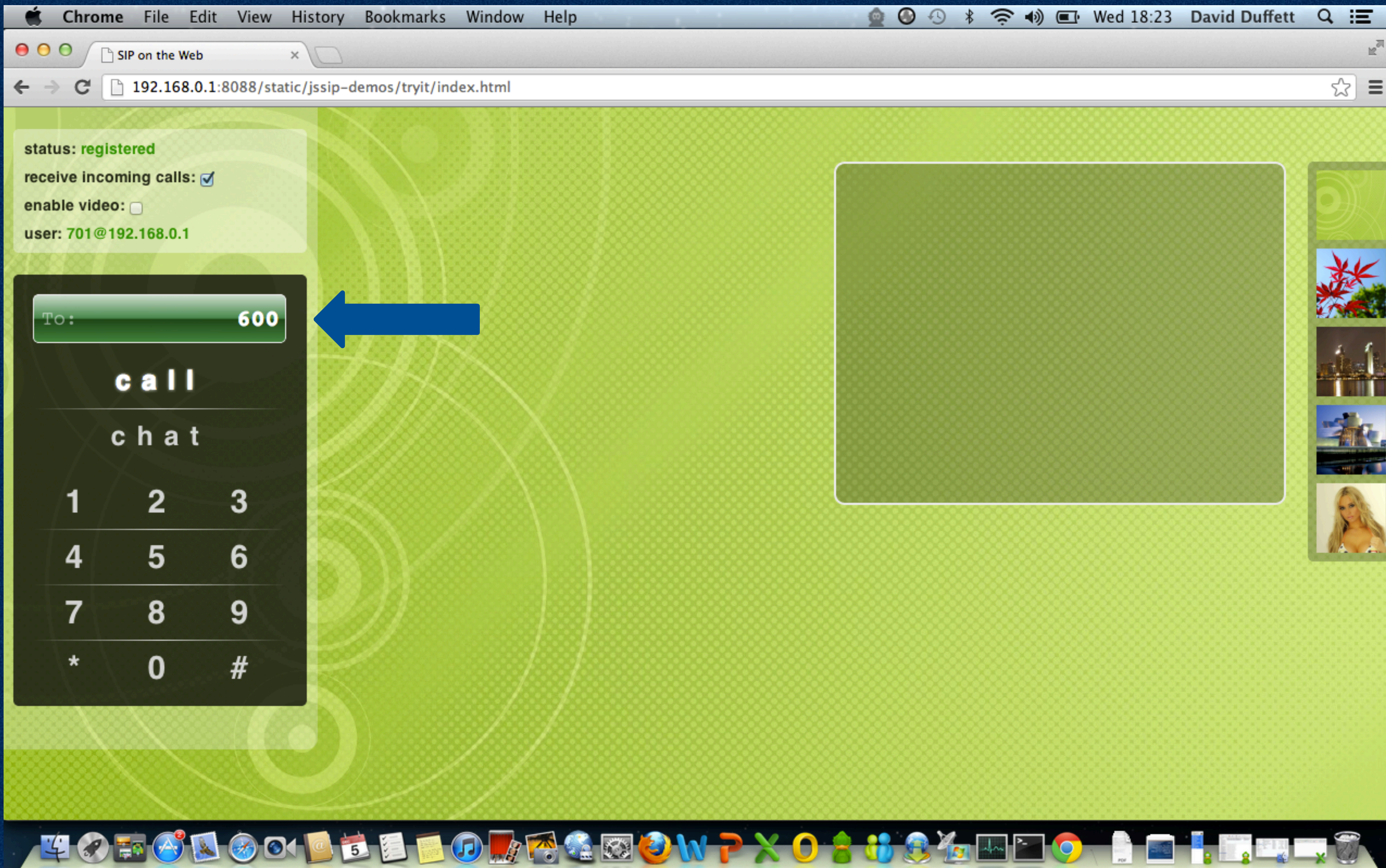


If successful

digium
The Asterisk Company

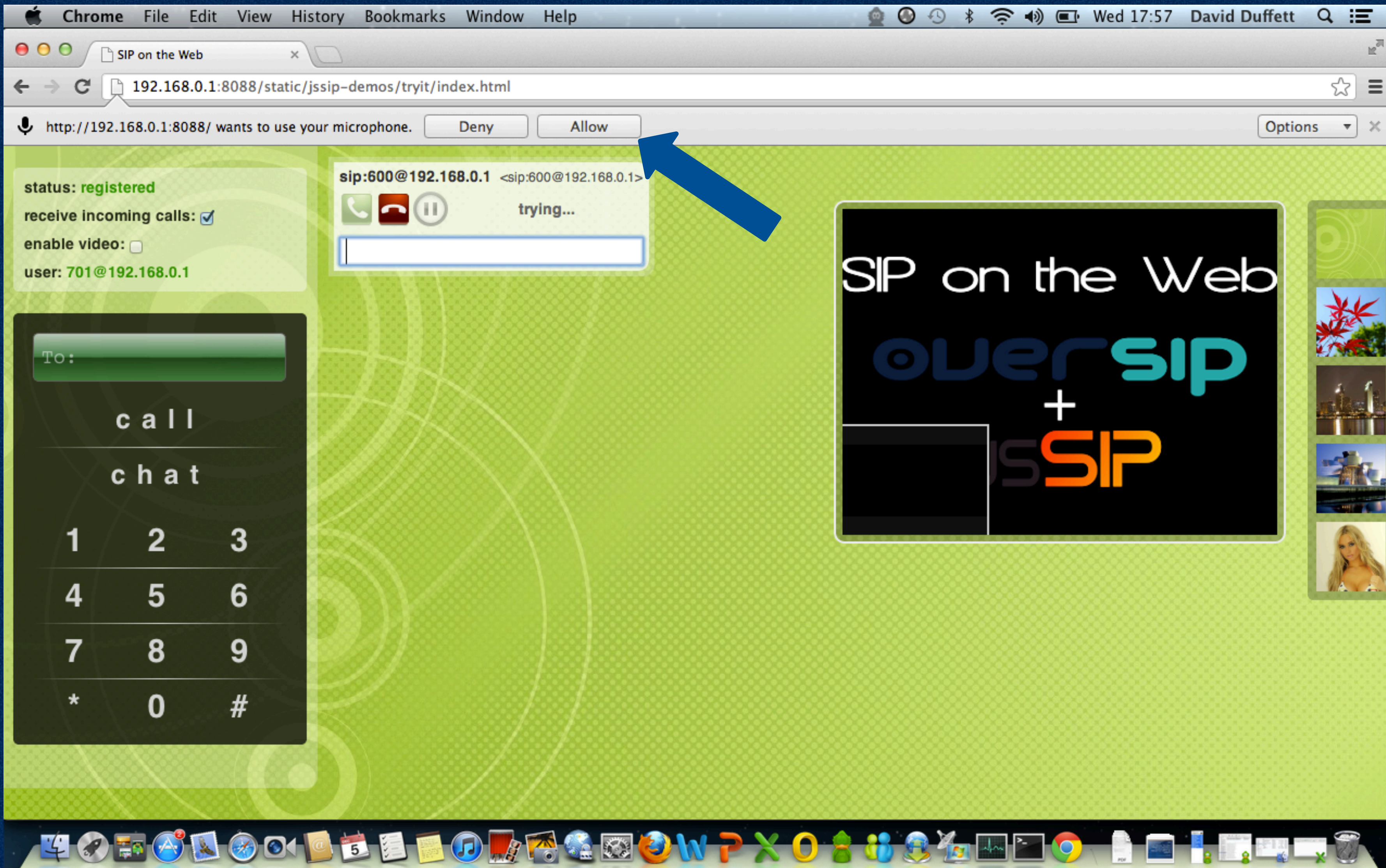


Put 600 in the box, and press <Enter>



Remember to click ‘Allow’

digium
The Asterisk Company



Creative Innovation – Customer Satisfaction – Continual Quality Improvement

If everything works…

digium
The Asterisk Company

- After a short pause, you should hear a special message
- From the lovely Allison – THE Voice of Asterisk

What questions do you have?

?

Summary

- Asterisk is 13 years old
- It is very established, and has a huge ecosystem around it
- Our technology and processes are maturing
- We’re still adding new functionality
- Asterisk 12 will be a real change in architecture to many of Asterisks subsystems, for the long term good of the project
- The community is very important in setting the agenda and doing some of the work.
- Let’s help each other to succeed!

Thanks!

dduffett@digium.com