R&E Telepresence Exchange status & lessons learned

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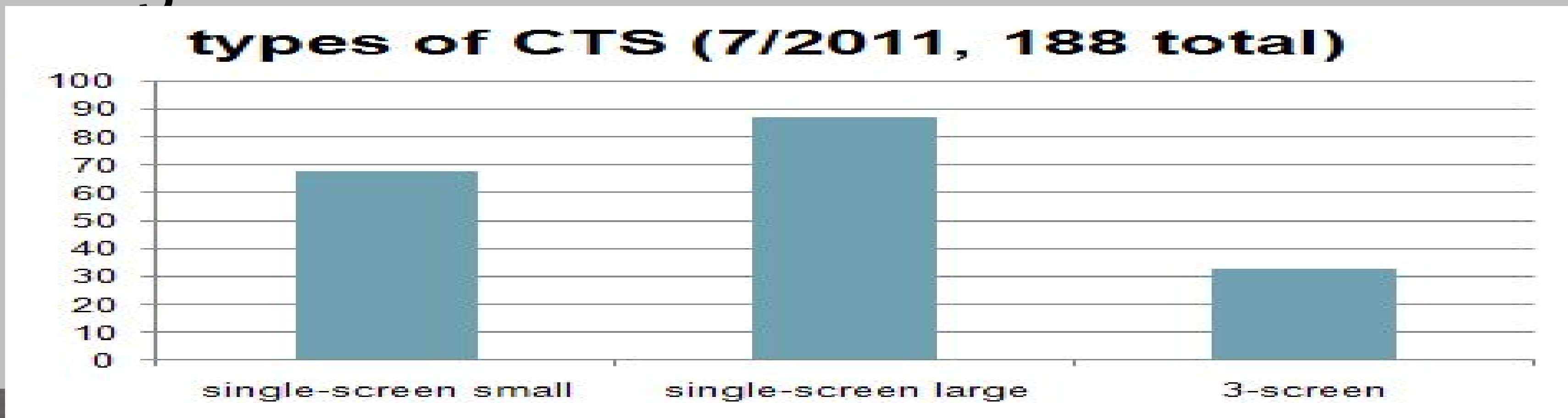
The R&E exchange community:
Cisco Telepresence rooms

- Currently almost 200 Cisco Telepresence rooms connected, many more ‘out there’
- About 80 institutions, most in US
- Single-screen and multi-screen



What is the R&E TP Exchange?

- Begun 2009, the central infrastructure that enables highly-functional, scalable, interconnection of many local, state/regional, and international telepresence systems
- Originally *Cisco* Telepresence, but not intended to be limited to Cisco only—now also:
- Interoperability gateway, standards-based interconnectivity to other SIP & H.323 devices

Where is the R&E TP exchange?

- The first one is in the center of *North America*
- AARnet has announced creation of a TP exchange for *Australia*
- Others should come at least in continents or major regions, especially:
  - *China*, either with or in addition to *Asian* exchange
  - *Europe*
  - *Latin America*
- *Federate* them, leveraging high-performance nets

What *is* that 'central infrastructure'?

- “SBC” Session border controller—performs call-admission, number analysis, call-routing, trunking
- Periodically monitors state of trunks via SIP *“OPTIONS ping*”, a kind of SIP 'hello‘
- Telepresence server blade, H.323 interop services
- Monitor quality of connections via Cisco IPSLA
  - Loss
  - Jitter
  - Latency

Central infrastructure 2

- Redundantly routed via NLR/Internet2 backbones
- Located in Kansas City in the NLR POP
- SIP trunk to each remote site or exchange
- Trunks to other exchanges (R&E + commercial)
- Interconnected with Internet2 backbone for reachability to Internet2 members
  - Exchanges limited routes with I2 for Telepresence
- Multipoint services: Telepresence multipoint switch (CTMS)

Central services
what do we do for user sites?

- Coordinate testing on turnup
- Coordinate R&E telepresence site directory
- Maintain mailing list for news, alerts, q&a, and website for FAQ & other information
- Represent community to vendors, providers
  - Enroll sites with commercial providers
- With community, help set standards
- *Not* end-user support, hardware support

Equipment at user end (1)

- Minimally:
  - Codec/screen/IP phone user interface
  - Cisco Call Manager (CM) for managing (up to many) endpoints, signaling, terminate trunk, call-routing, managing software, reporting, etc etc
    - CM function could be shared with other institutions

Equipment at user end (2)

- Optionally:
  - Redundancy, e.g. CM cluster
  - Local multipoint switch
  - Local interop options to other SIP or H.323 devices
  - firewall/border device(s)
  - Recording

Functional options for endsites

- NAT
- CTS-manager for scheduling, integration w/Outlook, can push calendar to phones
- Media (and/or signaling) encryption
- PSTN gateway

Endsite Requirements

1. Routed IP address(es) for Call Manager
2. Routed IP address(es) for codecs
3. E.164 “phone number” for codec: our standard is an ‘internationalized’ E.164 number correct for your locality. In North America, 11 digits: 1+(area code)+(exchange)+(local part)

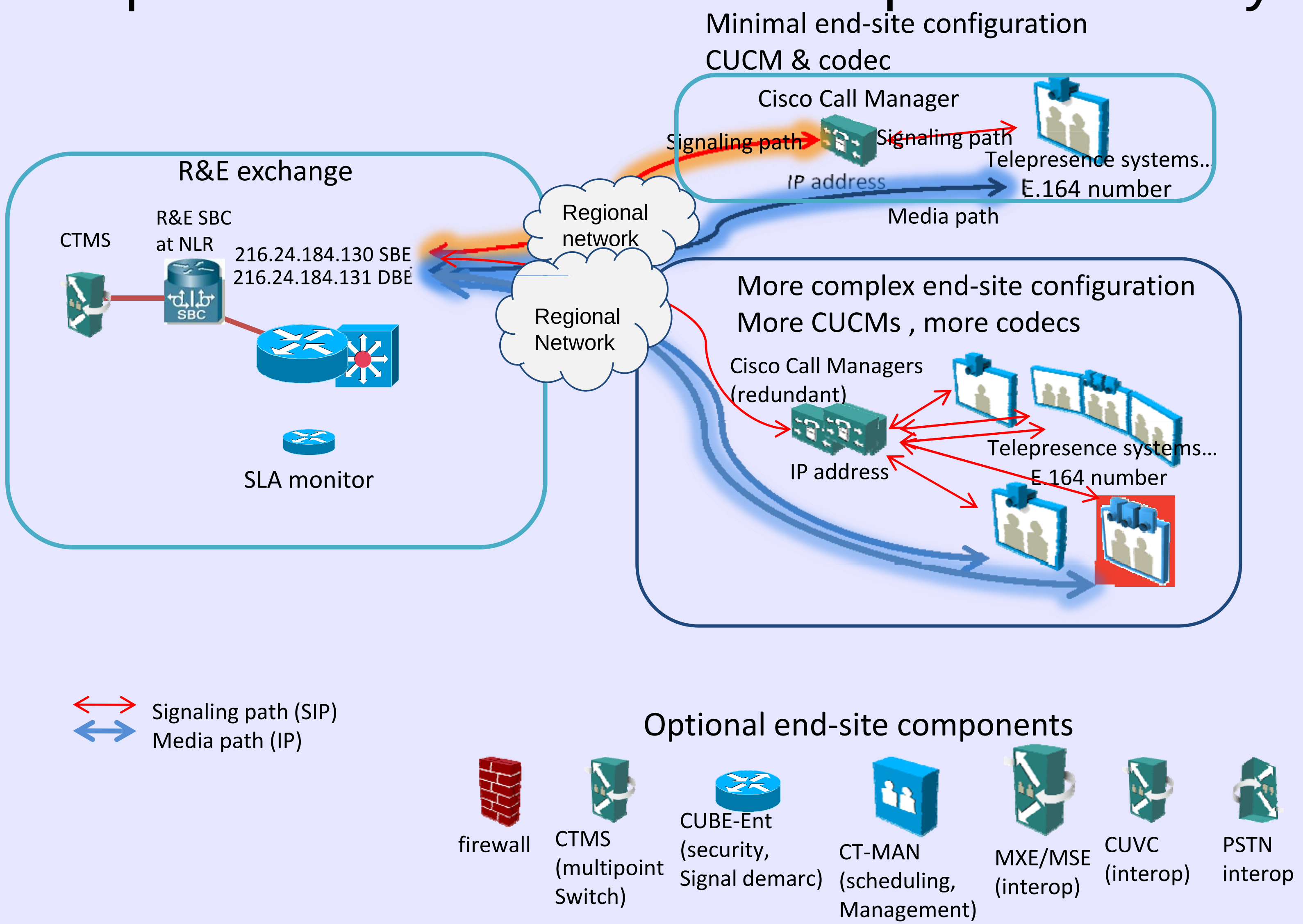
    For example (US) 1-919-123-4567 or (China) 86-1-21-12345

    CM understands international dialing

    Doesn’t <u>need</u> to be switchable; PSTN connection is optional

4. That number is your ‘dialing number’ outside, and you must answer when other sites call you with that.

Sample R&E TelePresence Components & Layout



What's needed to connect?

10,000-meter view:

You have a codec & call manager. To connect to & use the R&E exchange, you need:

1. IP reachability: A functional routed (layer3) connection that can reach the exchange
2. A SIP trunk to the exchange
3. A valid E.164 (phone) number & dial plan

Details #1: routed connection

- Traffic must be able to flow freely
  - All protocols are documented well
  - SIP signaling Call Manager⇔SBE 216.24.184.130
  - Media flows codec⇔DBE 216.24.184.131
  - Signaling on SIP port 5060/5061, media UDP RTP 16-32K
- Leverage existing high-performance networks
  - Only ~5Mbs/screen, no special circuits needed
- Traffic must be loss-free, low-latency, low-jitter

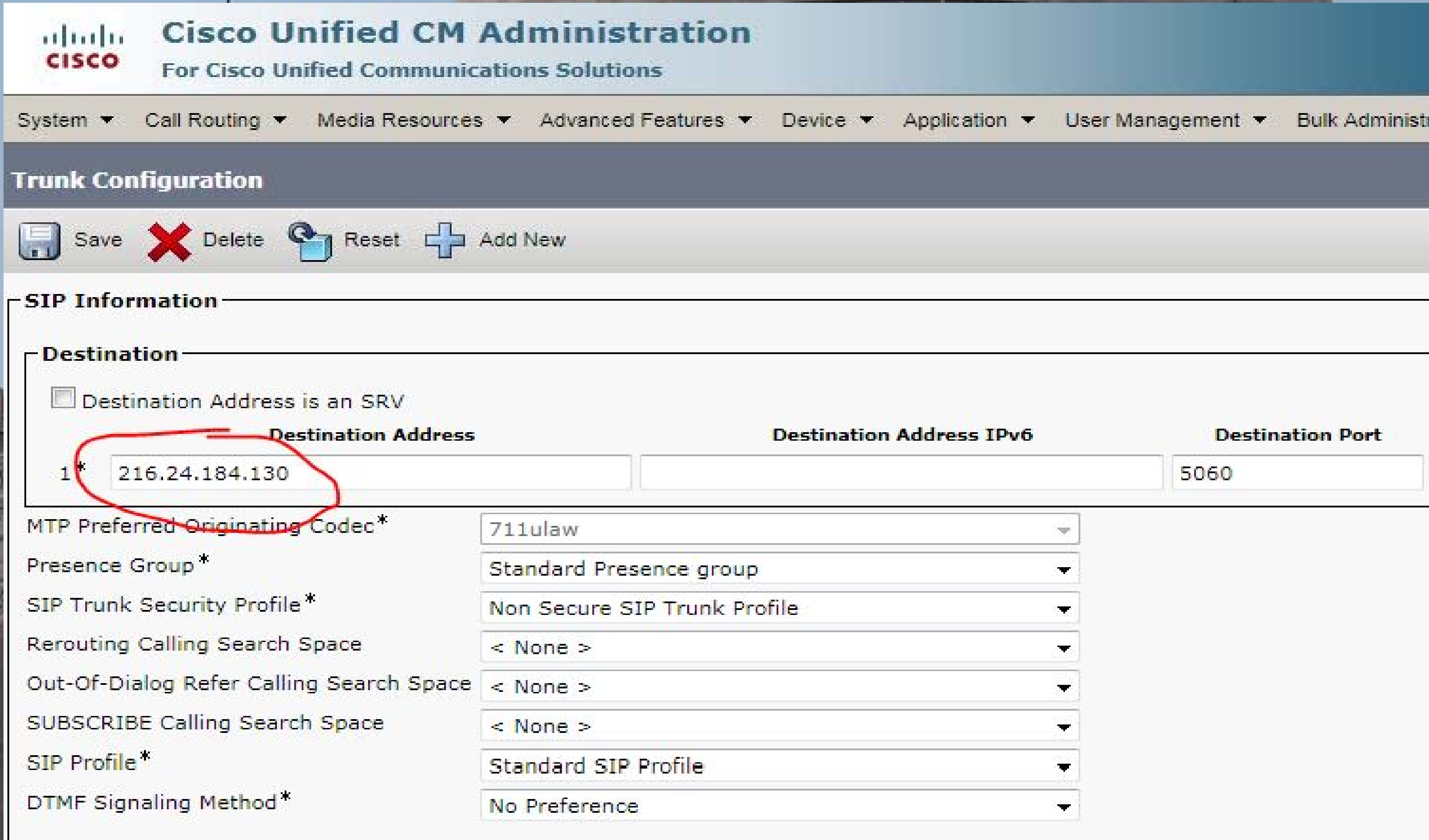
routed connection—*what can go wrong?*

- Firewall problems, for example letting signaling AND media through, or not getting enough SIP state. Sometimes the fix is to insert a ‘CUBE’ (proxy).
- NAT: ‘nuff said?
- Occasional special routing for non-members to get traffic to R&E exchange
- Loss, latency, jitter: jitter & latency issues are rare, but loss sometimes needs to be fixed with QoS. Bandwidth issues are very rare in our networks.

Details #2: SIP trunk

- Persistent SIP adjacency is created between CM and SBC by creating a SIP *trunk*
  - ❖ Uses IP addresses of each end
  - ❖ Since the trunk is stateless, the SBC periodically polls the CM over the trunk with an *OPTIONS* type of SIP packet to see if it answers. This hello-like interaction is called an ‘options ping’ though there’s no ICMP involved. The SBC can mark the adjacency as *online* or *offline* based on response.

Creating the SIP trunk (in CM)



SIP trunk—*what could go wrong?*

- If protocol path is opened correctly, this should work fine and almost always does.
- For (us) data people, SIP is generally a foreign language: how to decipher what *exactly* was wrong, or missing, in the negotiation?
- This is where we may see configuration issues with other parts of the CM or codecs, for example, wrong protocol or bandwidth settings.

Details #3: number & dial plan

- End site designates a valid E.164 number for each device (see our *standard* earlier)
- Number is programmed into the device via CM, associated w/ IP of known registered device
- Phone & codec are associated by virtue of same E.164
- CM may perform number manipulation on incoming or outgoing numbers …. more

Details #3: Dial plan (p.2)

- CM may have various trunks, dial-plan routes destination number (patterns) to trunks
  - Uses longest-match (most-specific) pattern
  - Knows all ‘local’ devices automatically
  - Generally punts everything else to exchange
  - Use North American (or other) Numbering Plan
  - Understands international dialing
  - So it’s possible to have a single dial-pattern: “@”

Dial plan—*what could go wrong?*

- One of the most frequent problems is that the CM uses a short version of the long phone numbers locally, and doesn’t recognize the full number when it comes in, refusing the call.
- Sometimes the CM doesn’t format the identification of the *outgoing* number correctly.
- Unnecessarily complex dial plans
- User confusion with TP, PSTN, local/LD prefix
- Wrong numbers—directory helps!

How does TP connection work?

‘above’ and ‘below’ the covers…

- Codec & phone register to Call Manager (CM)
- CM loads image & config (including directory, calendar) to codec & phone
- User dials (manually or via directory) number
- Codec signals call to CM (SIP)
- CM compares with dial plan, signals call to SBC
- …more

How does it work? #2

- SBC receives signaled call from CM
- SBC compares with dial plan, routes call to appropriate end-site trunk (incl interop sites)
- Remote CM receives signal, analyzes called number & call requirements if it wants to answer
- Remote CM signals orig CM (via SBC) that call is ok, state ‘active’, start to send media (via SBC)
- UDP Media begins to flow *<u>codec</u> to <u>SBC</u> to <u>codec</u>*

How does it work? #3
multipoint

- When >1 system is in call, use ‘multipoint switch’ (*CTMS*)
  - Just another SBC trunk, chosen by SBC’s dial-plan
  - No transcoding is necessary if all Cisco
  - Up to 48 screens at once, expanding to 90
  - Screen-switching, or site-switching
  - Supports encryption, blocking, listing, dial-out
  - All callers call the same E.164, CTMS joins them together
- Looks like a normal call

How does it work? #4
inter-exchange

- Inter-exchange calling uses the same fundamentals: IP connect, SIP trunk, dial plan
  - Usually need to create a new physical connection
  - Single or redundant trunks between exchanges
  - Dial plan selects correct trunk
  - Commercial exchanges don’t allow point-to-point dialing, only connect via their multipoint switches
    - Pro: Only one number for us to call for each
    - Cons: no p2p, no interop

Inter-exchange among R&E

- Some limitations in previous slide are not required for *technical* reasons: if participants are willing, they can be opened up:
  - Point-to-point calls across exchanges (likely more complicated dial plan)
  - Multipoint & Interop calls
  - calls
  - Directory services
  - etc

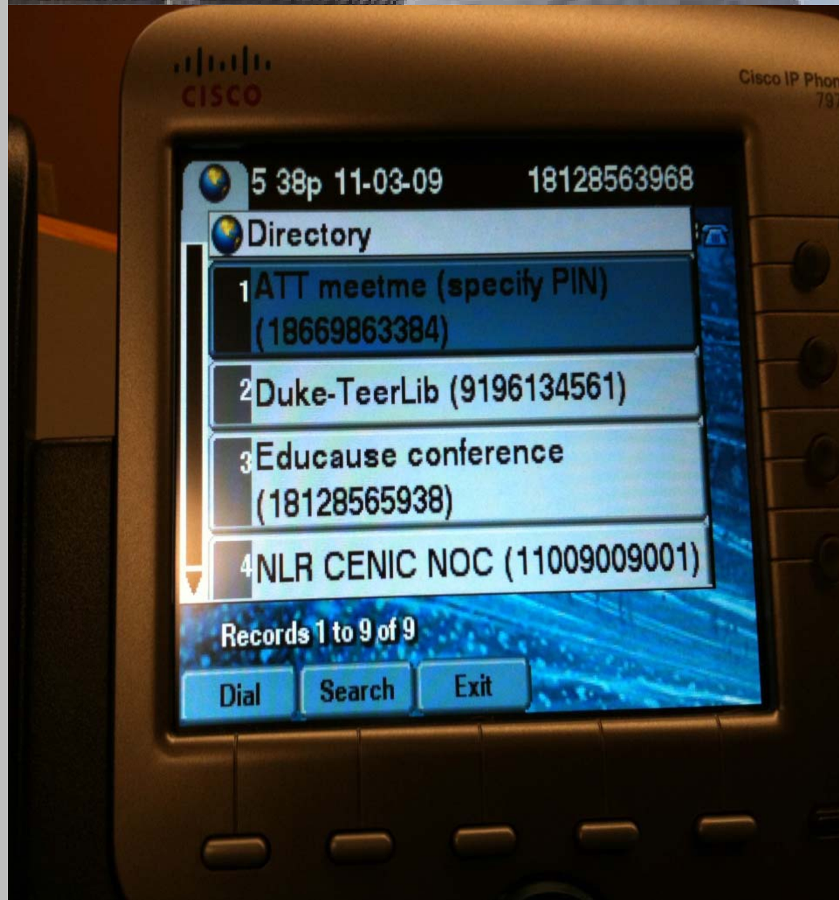
How does it work? #5

interop with SIP or H.323

- Generally requires a transcoding box from RadVision or, more recently, Codian->Cisco
- *Telepresence Interoperability Protocol*
- Starting summer 2011, new Cisco code allows direct p2p calls with endpoints that support “*H.264 baseline profile*” *standard* (needs version 8.6 in CM, 1.7.4 in codec)

Directory—locally

- Who? Where?
- There is an internal directory in the phone
- Populated from CM
- Same for all phones registered to that CM
- Can be created (CSV) & uploaded to CM
- Can have 100s of numbers



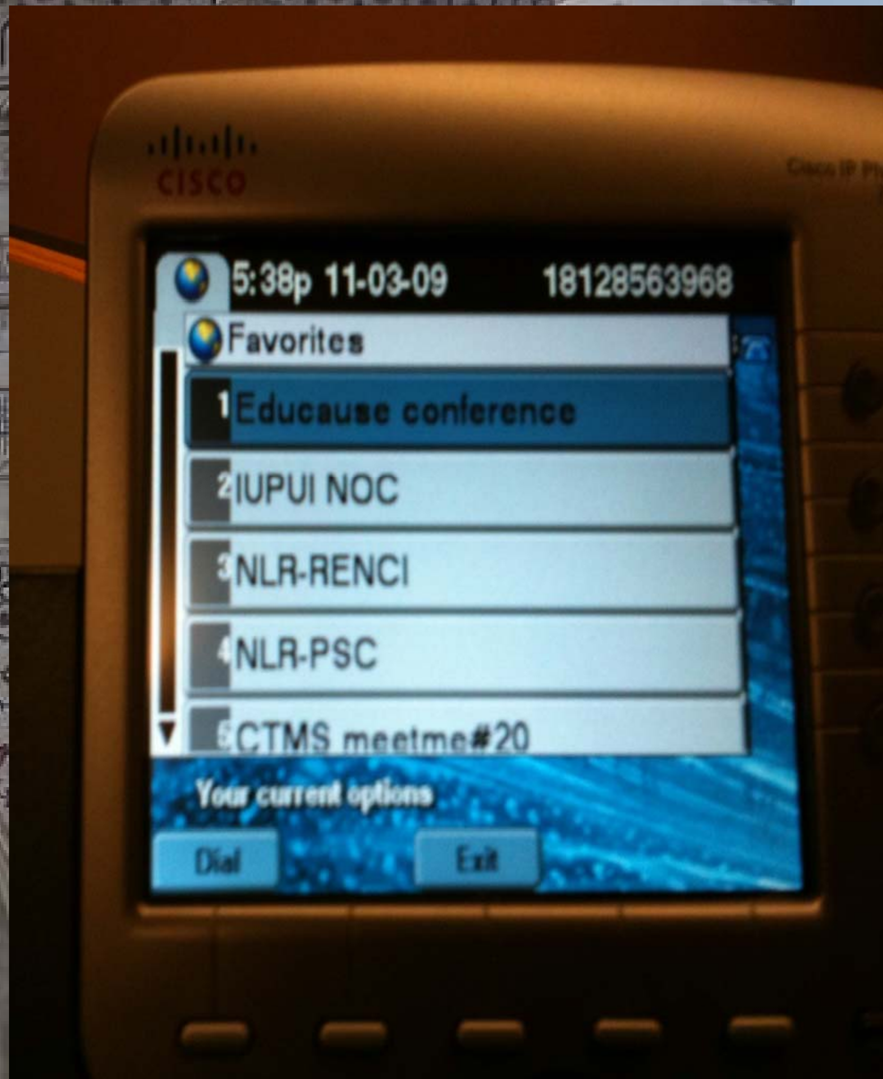
Directory—globally

- How do you find out what's out there, and where?
- How do you find who to talk with about it?
- What's its 'phone number'?
- As owner, how do you control access & visibility?
- How do you do these with PSTN or web today?

- R&E exchange directory
- North Carolina State University TP directory
- Cisco TP directory
- Commercial-provider directories
- Nothing global…
- No mechanism for auto-listing…

Phone ‘favorites’

- Configured from CM
- Appears on IP phone
- Different for <u>each</u> phone



Calendaring

Telepresence room as a resource

- Two parts—think of each for inside/outside user:
  - See availability
  - Commit availability
- How broad a view is appropriate?
  - Can you schedule someone else’s resources?
  - *Should* you be able to? (A&A issue)
  - Should you be able to see if/when they’re available?

Calendaring p.2—*intra-enterprise*

- Can schedule *your* devices (codecs, CTMS) as resources (mail, web) with *CTS-man* appliance, integrating resource reservation nicely with phone itself
- CTS-man can connect to groupware calendar/ resource-mgt app (e.g. Outlook), or other apps via API
- CTS-man pushes calendar to phone, ‘one button’ call
- No good <u>inter</u>-enterprise solution today (API?)
- No ‘open’ way to push calendar to phone/CM

IPv6 and Cisco Telepresence

- Cisco supports end-to-end IPv6 *VOIP* calls
  - ❖ SIP Signaling works over native IPv6
  - ❖ Registration of devices works over IPv6
  - ❖ VOIP media flows end-to-end over native IPv6
- Cisco does not *yet* support v6 for telepresence calls (media support still missing)
  - ❖ R&E community among leaders in asking for this

For more information

Noc.nlr.net pages on Telepresence, including FAQ & map of connected sites

- Noc.nlr.net > Documentation > Telepresence: information on many aspects of connection and maintenance of your connection, including:
  - Dial plan information & instructions
  - List of connected endsites
  - How-tos, configuration guides
  - GRNOC router proxy gives you visibility

Thank you!

Credit for this photo: Wikipedia (‘India’)