

SIP-based VoIP Deployment in Taiwan

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Outline

- Introduction to TWAREN
- NTP SIP-based VoIP Platform
- Plans of VoIP Working Group
- Prototypes of Some Utilities








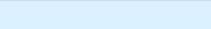
TWANREN

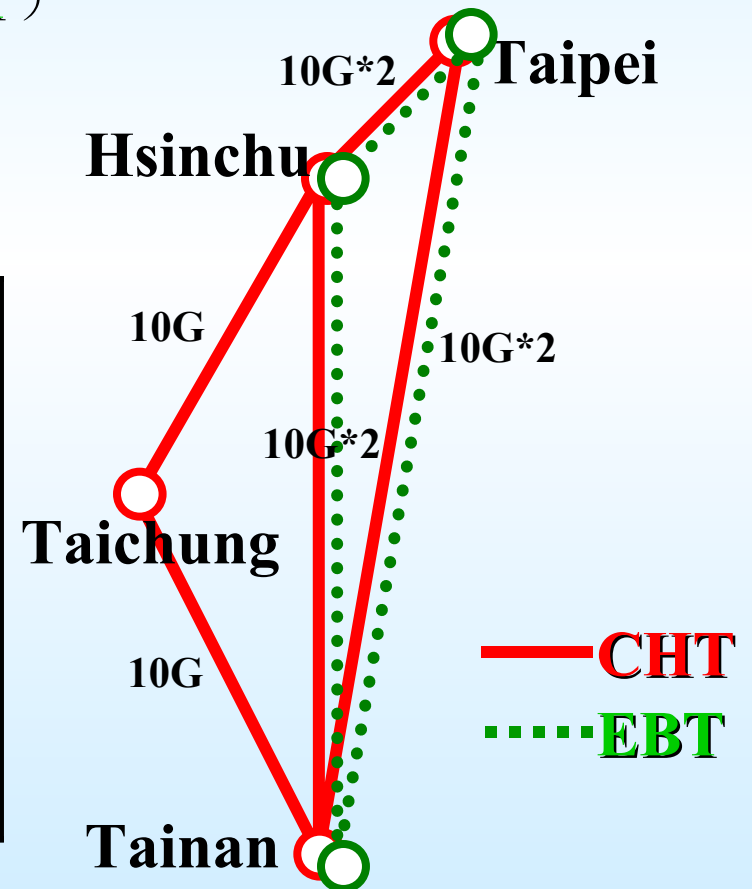
- TWAREN - TaiWan Advanced Research & Education Network
 - <http://www.twaren.net/English/index.htm>
- Dual physical circuits & Three network systems
 - Production Network
 - Provide common academic usage
 - Provide usual utility
 - Research Network
 - Provide advanced tech. (IPv6, MPLS, Multicast...)
 - Backup with Production Network
 - Optical Network
 - Provide layer1 provisioning

Backbone Network

- **TWAREN Backbone Network topology**
(Provided by dual carriers **CHT**, **EBT**)

Bandwidth of each link is 10 G

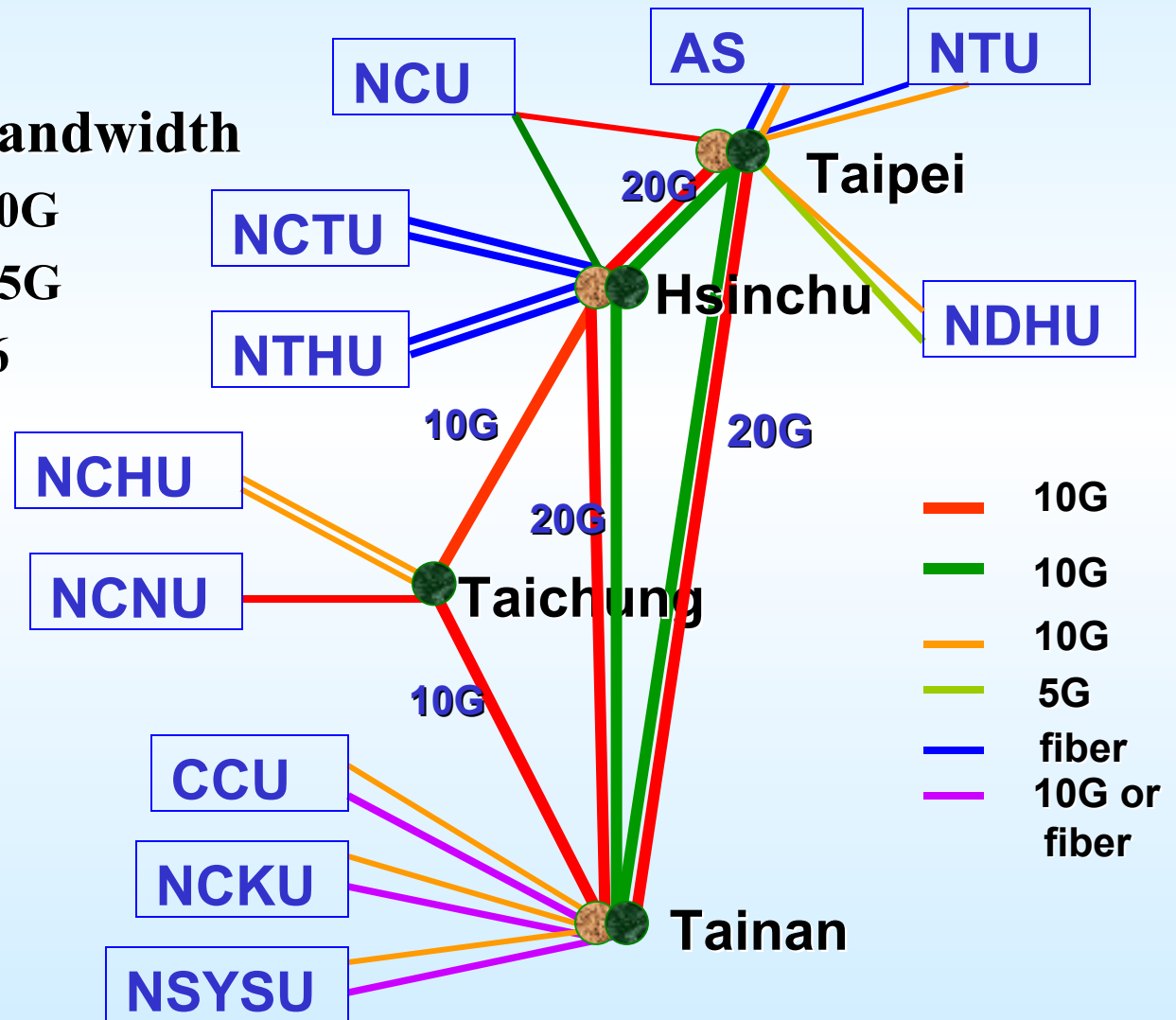
Core		Core
Taipei	 	Hsinchu
Taipei	 	Tainan
Hsinchu	 	Tainan
TaiChung		Tainan
Hsinchu		TaiChung



POP (Point of Presence)

- **Aggregated bandwidth**

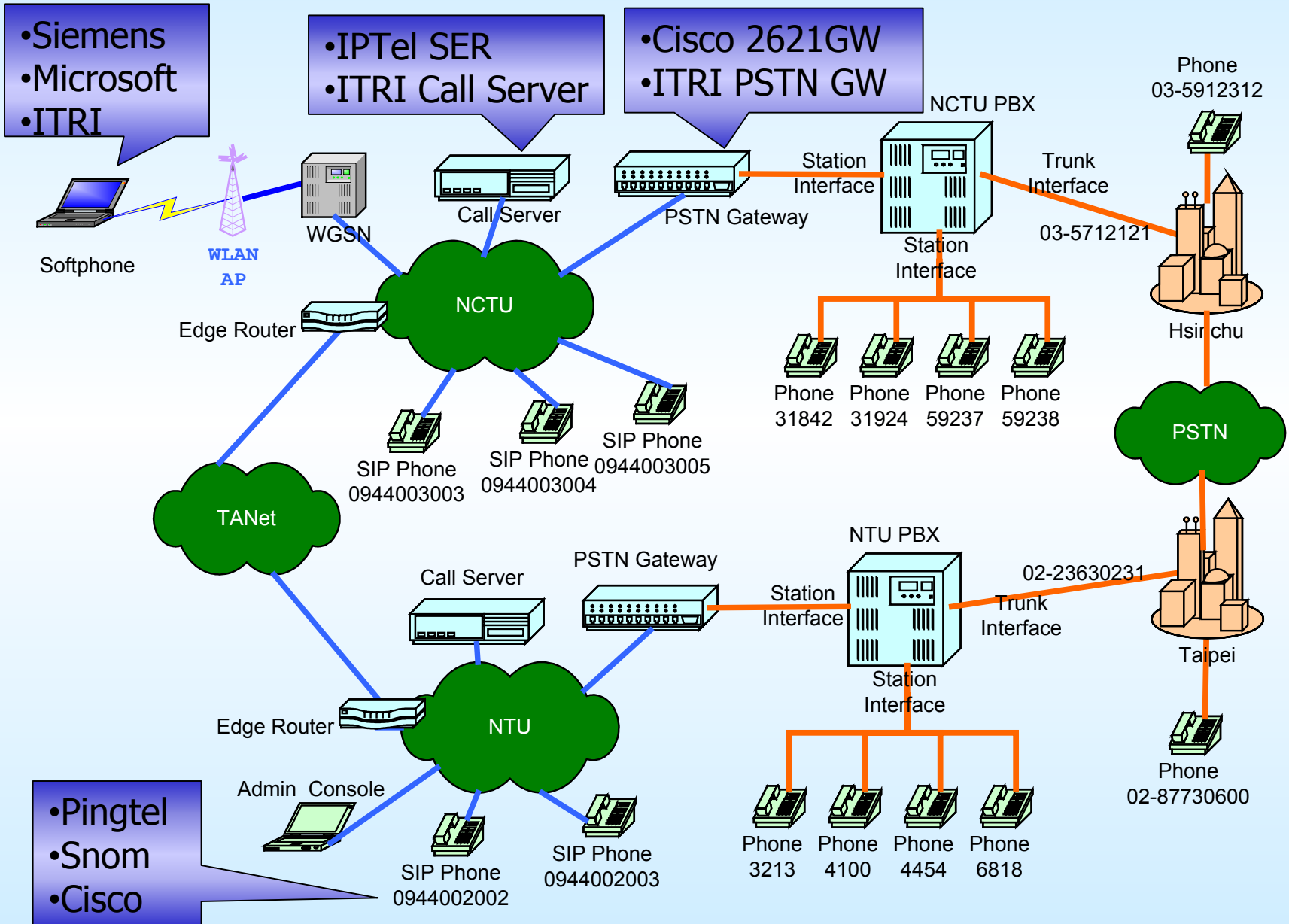
- Backbone: 80G
- Regional: 145G
- Dark fiber: 6



VoIP on TWAREN

- Why should TWAREN promote VoIP
 - VoIP is convenient.
 - VoIP to Internet2 schools are free.
 - VoIP has hot research topics.
 - VoIP enables rich services.
- How should TWAREN promote VoIP
 - TWAREN has good QoS infrastructure.
 - TWAREN supports end-to-end performance measurement.
 - TWAREN runs a conference bridge.
 - TWAREN provides a transition mechanism from H.323 to SIP.

NTP VoIP Platform

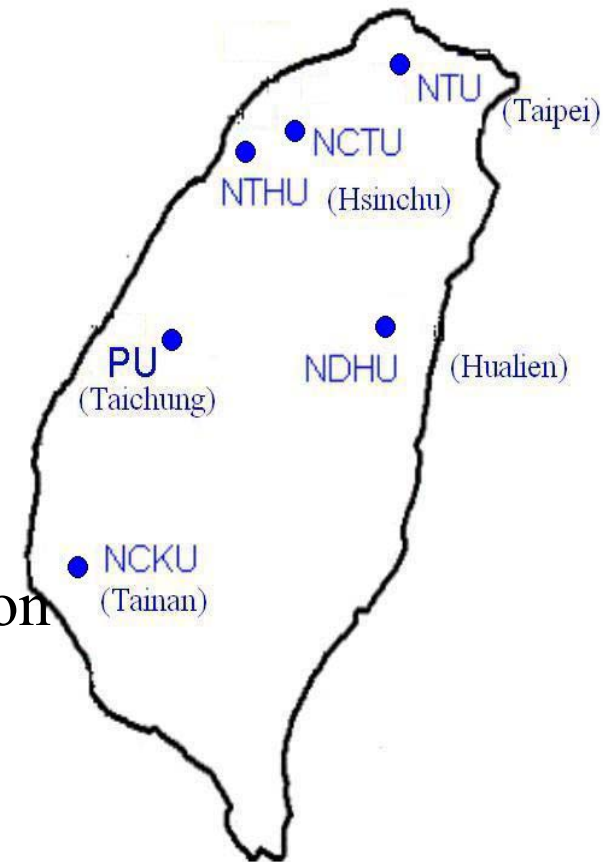


Academic Researches

Support academic researches on NTP VoIP Platform



- **NTU**: SIP Signaling Performance Evaluation on SCTP
- **NTHU**: Secure RTP and Location Privacy on VoIP System
- **NDHU**: Voice over IP study on All IP networks
- **NCKU**: DNS/ENUM Automatic Updating Mechanism
- **NCTU**: NAT Traversal & WGSN Project for Integrated Wireless VoIP Services



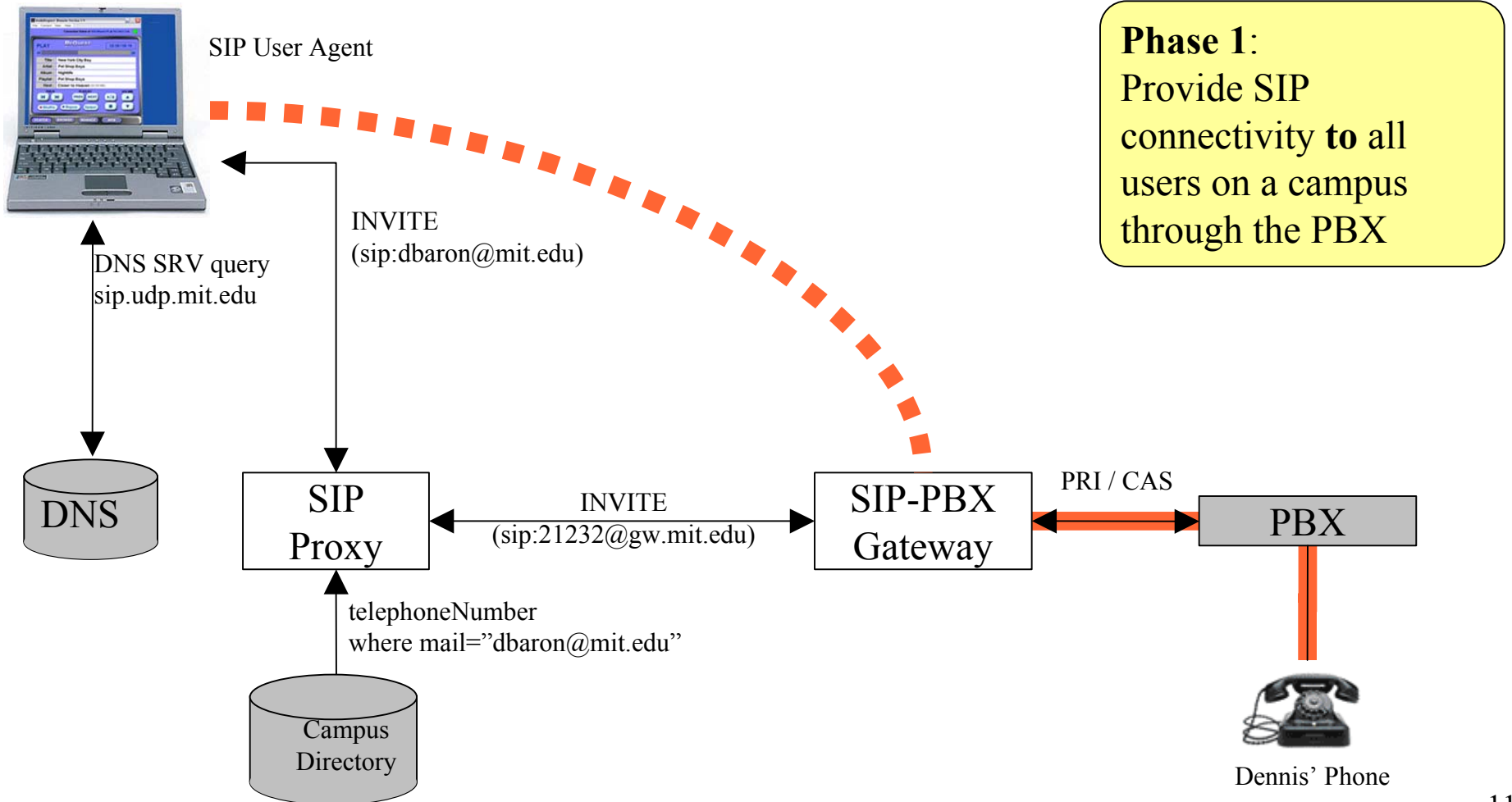
Numbering Plan

- GDS (Global Dialing Scheme)
 - 886-3-5712121-59238
- SIP URI
 - sip:solomon@ipv6.club.tw
- ENUM
 - 0944020678
- *"A rose by any other name would smell as sweet."*
 - William Shakespeare

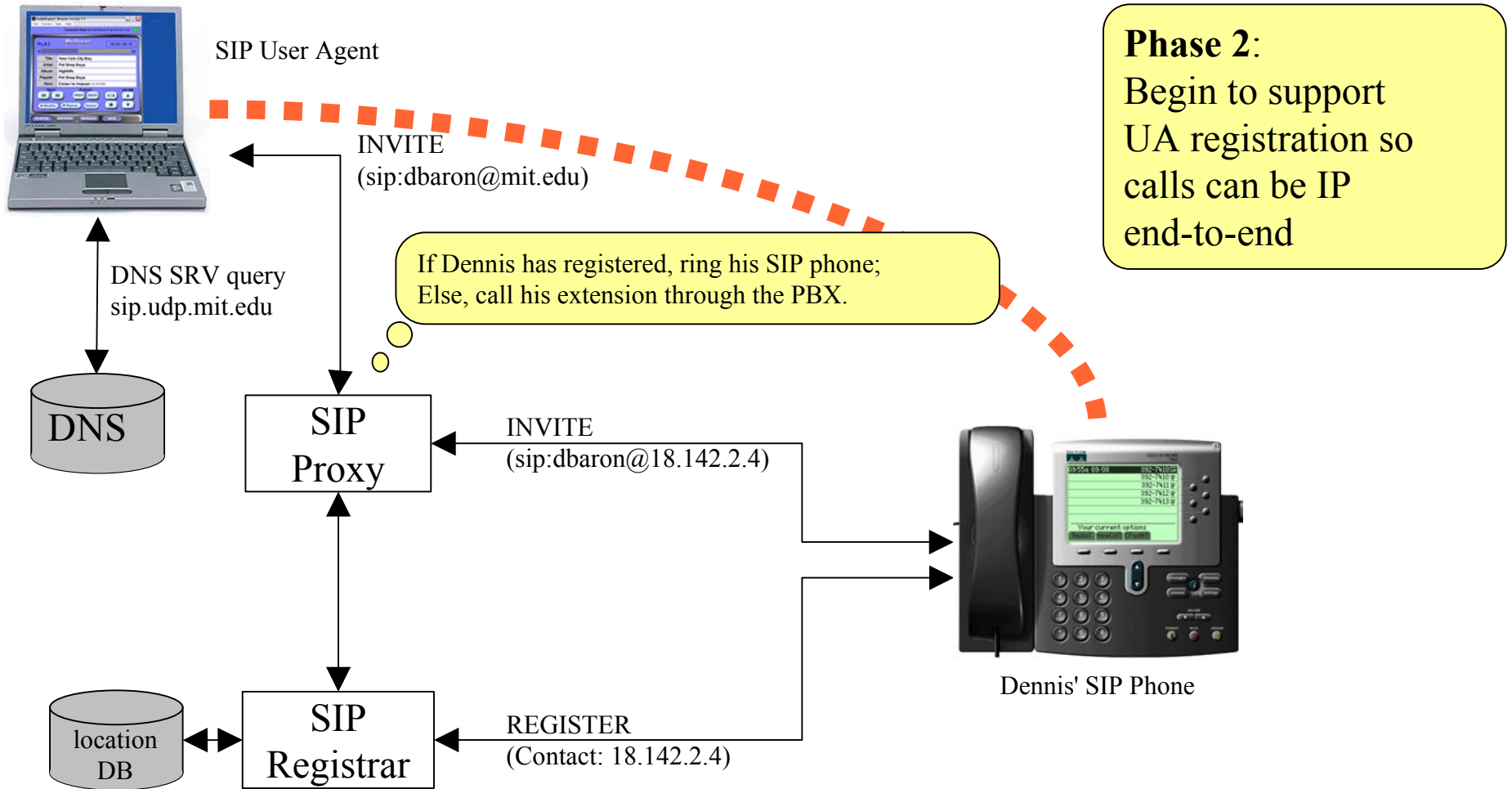
TWAREN VoIP Working Group

- TWAREN is chartering a VoIP WG.
- Proposed projects in 2004 includes:
 - SIP.edu
 - SIP/H323 Gateway + Conference Bridge
 - E2E Performance Measurement + Trouble-Ticket System
 - NAT Traversal (STUN, TURN, UPnP, IPv6)
 - Instant Message & Presence Service
 - BoD for VoIP

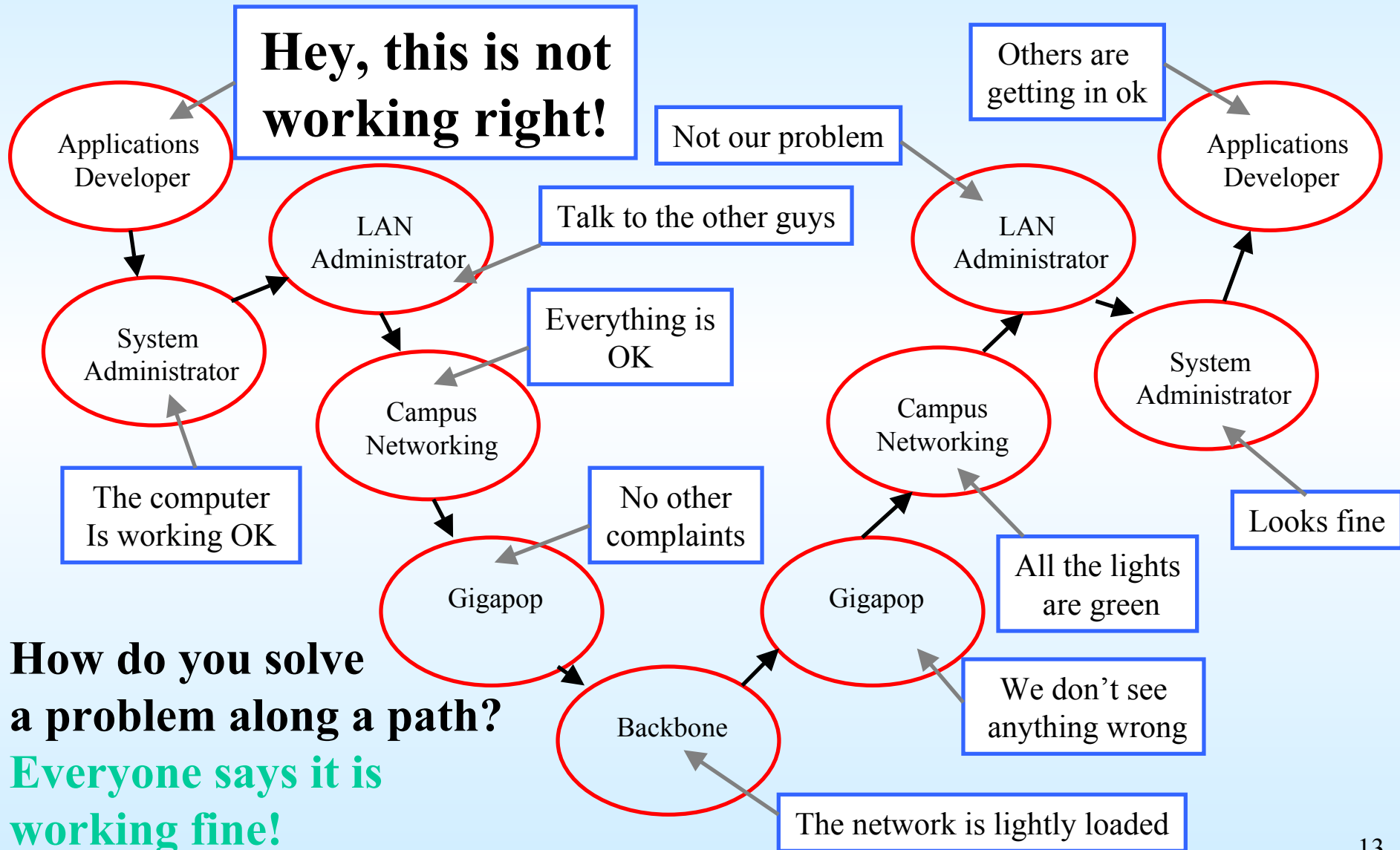
SIP.edu – Phase 1



SIP.edu – Phase 2



E2E Problems



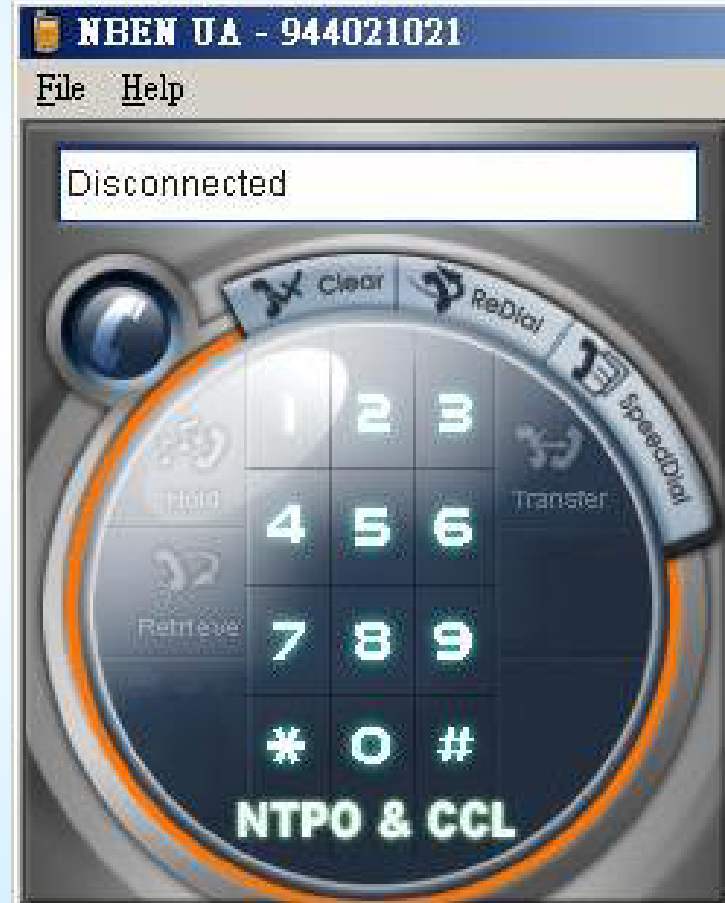
How do you solve a problem along a path?
Everyone says it is working fine!

Software Under Development

1. SIP UA with NAT Traversal
2. IPv6 SIP UA
3. IPv6 SIP Packet Analyzer

Project 1: SIP User Agent for NAT Traversal

- NBEN UA runs on Windows 2000/XP/2003.
- Both signaling and media data are transported on UDP.
 - SIP: port 5060
 - RTP: port 9000
- Support audio codec:
 - G.711 (64Kbps)
 - G.729 (8Kbps)
 - G.723.1 (6.3Kbps)
- Support STUN (RFC 3489) for NAT traversal.



Project 2: IPv6 SIP UA



Enter SipURL to dial

[sip:[2001:238:f88:131:202:b3ff:feca:60e5]:5060] Dial List Cancel

Preferences

Codec Settings | Credential Setting | RTP Settings

User Settings | Server Settings

User Information

Display Name : VoIPUser

User Name : ual

Public Address : sip:ual@[2001:238:f88:131:20c:6eff:fe49:1b98]

Contact Address : sip:ual@[2001:238:f88:131:20c:6eff:fe49:1b98]

Local Settings

Use IP Address : 2001:238:f88:131:20c:6eff:fe49:1b98

Static Assign 2001:238:f88:131:20c:6eff:fe49:1b98
2001:238:f88:131:f5f0:27bd:c9f0:69b8
fe80::1%1

UA Port 5060 UDP TCP

確定 取消 套用(A) 說明

Project 3: IPv6 SIP Analyzer

SIP Packets Capturing

SIP Signaling Flow

The screenshot displays the SIP Analyzer V0.1 interface, which is divided into several functional areas:

- SIP Packets Capturing:** The top-left pane shows a 'Session List' table and a 'SIP Packet List' table. The 'SIP Packet List' contains the following data:

No.	Time	Direction	Type	Destination
167	18:29:53(453518)	SIP Request	INVITE	sip:ua1@120...
169	18:29:53(467824)	SIP Response	SIP/2.0 100 Tryi	-
169	18:29:53(520162)	SIP Response	SIP/2.0 180 Ring	-
176	18:29:55(819430)	SIP Response	SIP/2.0 200 OK	-
178	18:29:56(319393)	SIP Response	SIP/2.0 200 OK	-
210	18:29:57(319397)	SIP Response	SIP/2.0 200 OK	-
232	18:29:57(896202)	SIP Request	ACK	sip:ua1@120...
- SIP Signaling Flow:** The top-right pane displays a sequence diagram of SIP messages:

```
INVITE sip:ua1@[2001:238f88:131:20c:6eff:fe49:1b4a]:5060 SIP/2.0
CSeq: 1 INVITE
SIP/2.0 100 Trying
CSeq: 1 INVITE
SIP/2.0 180 Ringing
CSeq: 1 INVITE
SIP/2.0 200 OK
CSeq: 1 INVITE
```
- Traffic Statistics:** The bottom-left pane shows 'Host Traffic' and 'Protocol Distribution'. The 'Protocol Distribution' pie chart shows the following data:

Protocol	Count
Other	5,097
IPv4	3,485
IPv6	1,495
- RTP Monitor & Playback:** The bottom-right pane shows an 'RTP Spy' window with a 'Session List' table:

Session	SSRC	Media Type	Packet C...	Length(s...)
140.113.2...	0(0x0)	11(0xb) ...	37	
140.113.2...	0(0x0)	55(0x37)	1	
140.113.2...	13467182...	0(0x0) -...	1	
140.113.2...	1(0x1)	5(0x5) -...	12	
140.113.2...	29626(0x...	0(0x0) -...	2060	40

Traffic Statistics

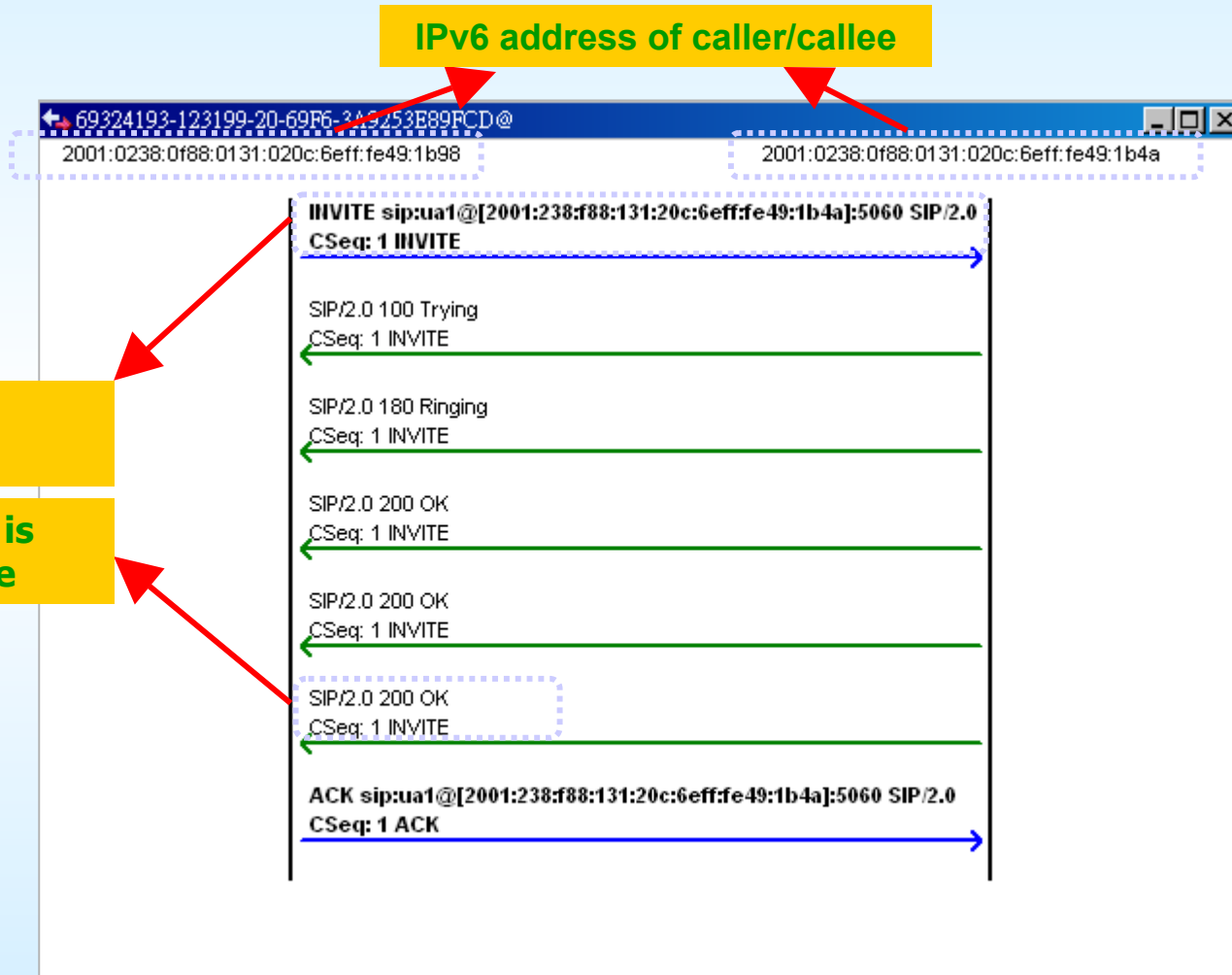
RTP Monitor & Playback

SIP Signaling Flow (1)

IPv6 address of caller/callee

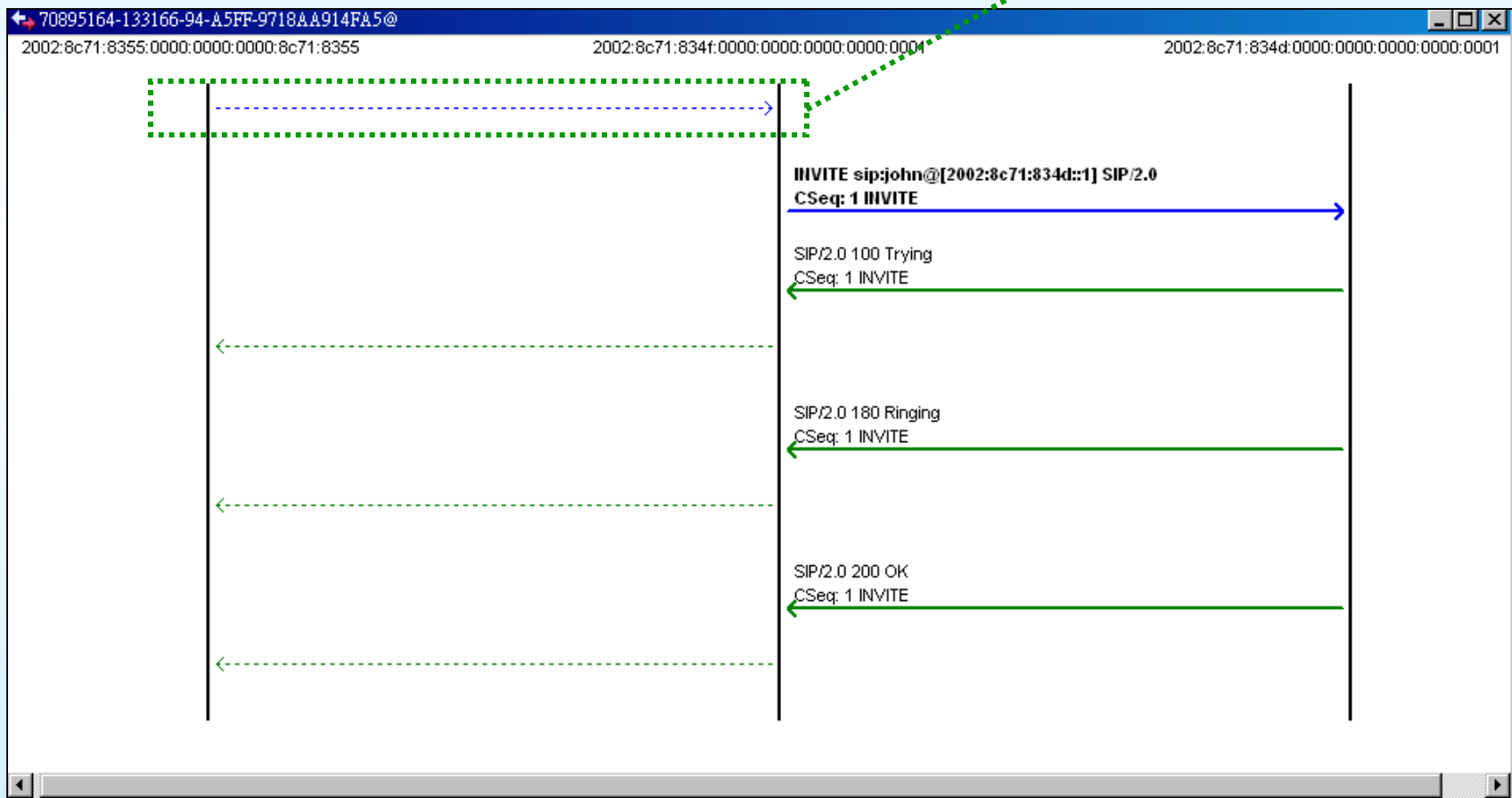
Blue arrow is
SIP Request

Green arrow is
SIP Response



SIP Signaling Flow (2)

Dashed arrow represent a conjectured signal (according to the Via/Route header field)



RTP Monitor & Playback

- Original purpose is to help assessing the packet loss rate of RTP traffic.
- It turns out to a tool to demonstrate the importance of encryption.

The screenshot shows the RTP Spy application interface. The title bar indicates the file path: D:\demo_packet\ipv6_h263_pcmu.cap. The application has a toolbar with various icons. Below the toolbar is a tab labeled 'RTP Spy'. The main area is divided into two sections: 'Session List' and 'Media Instance'. The 'Session List' section contains a table with columns: Session, SSRC, Media Type, Packet ..., and Length(secs). The 'Media Instance' section contains a table with columns: Media Description, Status, Packet Count, and Length(secs). At the bottom left, there are playback controls (stop, play, pause) and a status bar showing 'Parsed : 13110'.

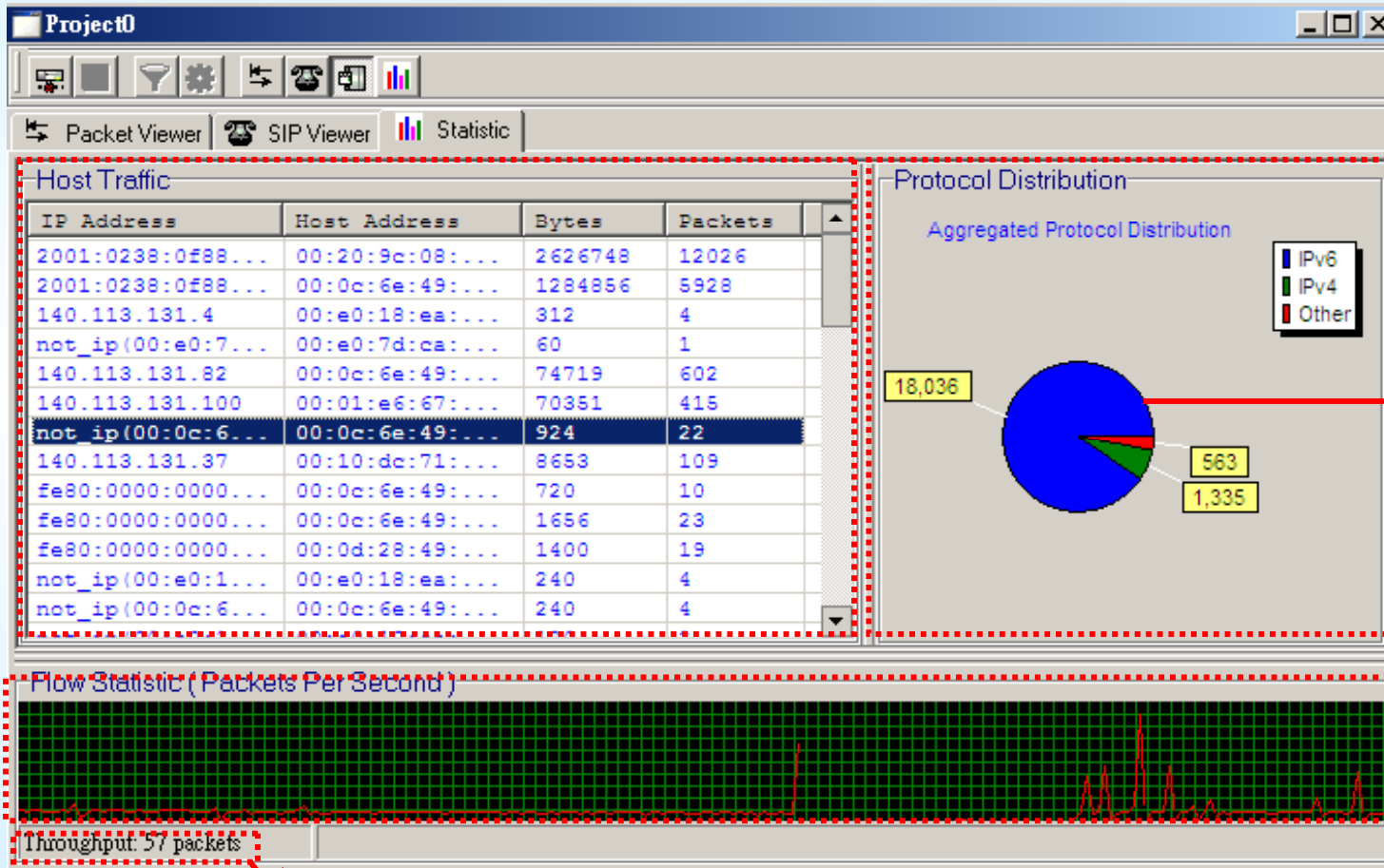
Session	SSRC	Media Type	Packet ...	Length(secs)
2001:0238:0f88:0131:451d:cbd0:1d4e:eb36:9000	2396(0x95c)	0(0x0) - PCMU(8...	988	19
2001:0238:0f88:0131:5c4e:c956:a0e3:5bd9:9000	30459(0x7...	0(0x0) - PCMU(8...	6392	127
2001:0238:0f88:0131:451d:cbd0:1d4e:eb36:9002	9395(0x24b3)	34(0x22) - H.26...	1984	
2001:0238:0f88:0131:5c4e:c956:a0e3:5bd9:9002	207(0xcf)	34(0x22) - H.26...	2073	
140.113.131.127:137	0(0x0)	64(0x40)	4	

Media Description	Status	Packet Count	Length(secs)
2001:0238:0f88:0131:5c...	Playing(279)	1447	28
2001:0238:0f88:0131:45...	Ready	580	11

RTP Streams

RTP Stream Playback

Statistics Data



Throughput (packet/s)

Conclusion

- By establishing a nation-wide VoIP testbed, TWAREN wishes to promote the convergence of voice and data services and encourage advanced researches in Taiwan.
- SIP coverage in 2003 is approximately 50,000 users. NTP plans to double the coverage in 2004.
- There are prototypes of NAT traversal solutions and IPv6 clients. Larger deployment is needed to verify these techniques.
- VoIP WG needs to closely work with Measurement WG and Multimedia WG to leverage our efforts. It is also critical to consolidate our on-going projects in accordance with Internet2 VoIP Working Group.