Overview of HATS

Content:

1. HATS General: This slide introduces the HATS activities to ensure the interconnectivity among various Info-communication equipment in multi-carrier/vendor environment of Japan.

2. SIP VoIP Interconnection

3. MPEG4, H.264 Interconnection

Sony co.

Masashi Tonomura

29 October, 2007

1. HATS General -What is HATS ?-

HATS Conference: Promotion Conference of Harmonization of Advanced Telecommunication Systems

Activates to assure interconnectivity and interoperability between info-communication equipment of different manufacturers

HATS is the Non-Profit organization to ensure the Telecommunication Equipments Inter-operability in order to give a user convenience.

HATS was established in Aug. 1988.

- Members: info-communications manufacturers,vendors, carriers, TTC, MIC[*1]
- Secretariat: Communications and Information network Association of Japan (CIAJ)

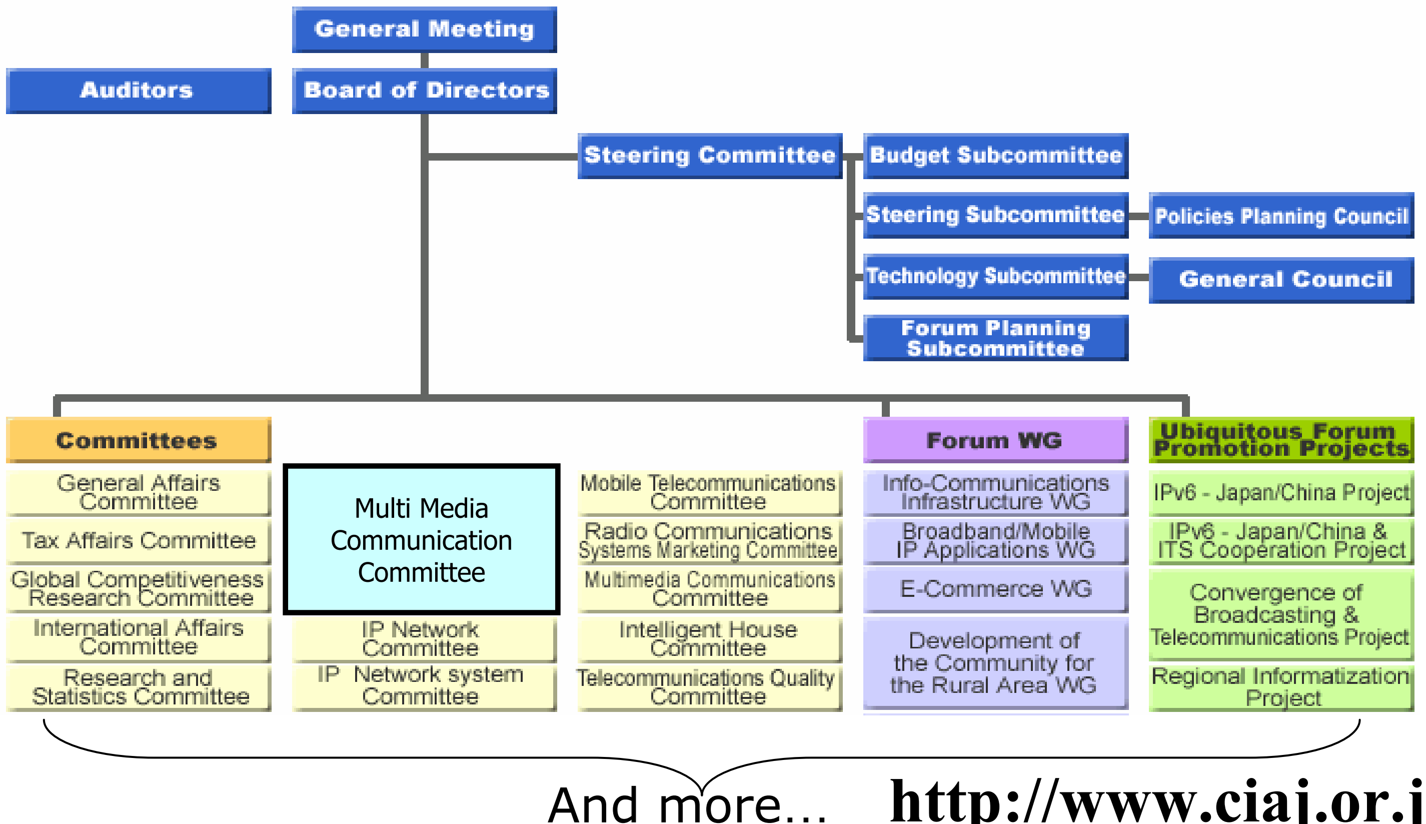
Note*1:MIC(Ministry of Internal Affairs and Communication, At that time MPT: **Ministry of Posts and Telecommunications**

For the details,
http://www.ciaj.or.jp/hats/e/what/about.html

What is the CIAJ ?

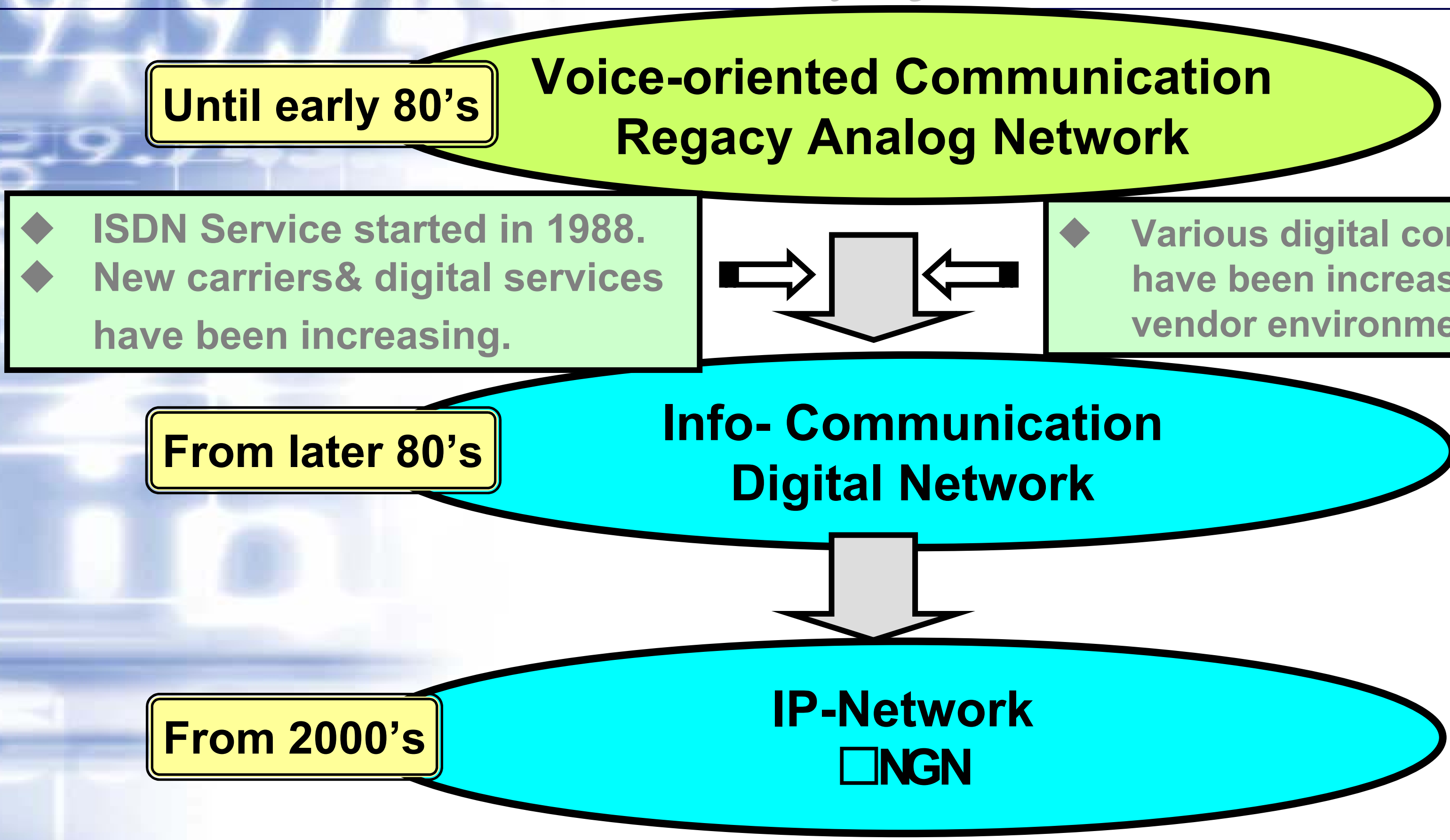
CIAJ: Communications and information network association of Japan
CIAJ is committed to the healthy development of info-communication network industries through the promotion of info-communication technologies (ICT) in Japan.

The Multi Media Communication Committee discuss the technical issues about multi media communication equipment, and act the IOT with HATS which is a group of CIAJ.



http://www.ciaj.or.jp/e.htm

Necessity of HATS



<Interconnectivity of communication systems is required>

In order to develop sound Info-communication market, a framework aiming at ensuring end-to-end interconnectivity among various Info-communication equipments was needed in multi-carrier/multi-vendor environment.

Structure of HATS Conference

Chairman

Executive Committee

-decides basic policies, overall HATS activities and establishment of new TILC

Steering Committee

-researches on ICT standardization & market trends
- coordinates TILCs activities
- liaises with other organizations

Advisory Committee

- gives advice on overall HATS activities

Promotion Committee & Demonstration Committee

- disseminates HATS activities
- plans/arranges seminars & demonstrations

Test Implementation Liaison Committees(TILCs)

-Plans & implements interconnectivity test
- examines test method/procedure
- reviews&studies test result

Telephone & TA TILC

Facsimile TILC

PBX Telecom ServerTILC*1

Inter-LAN TILC*2

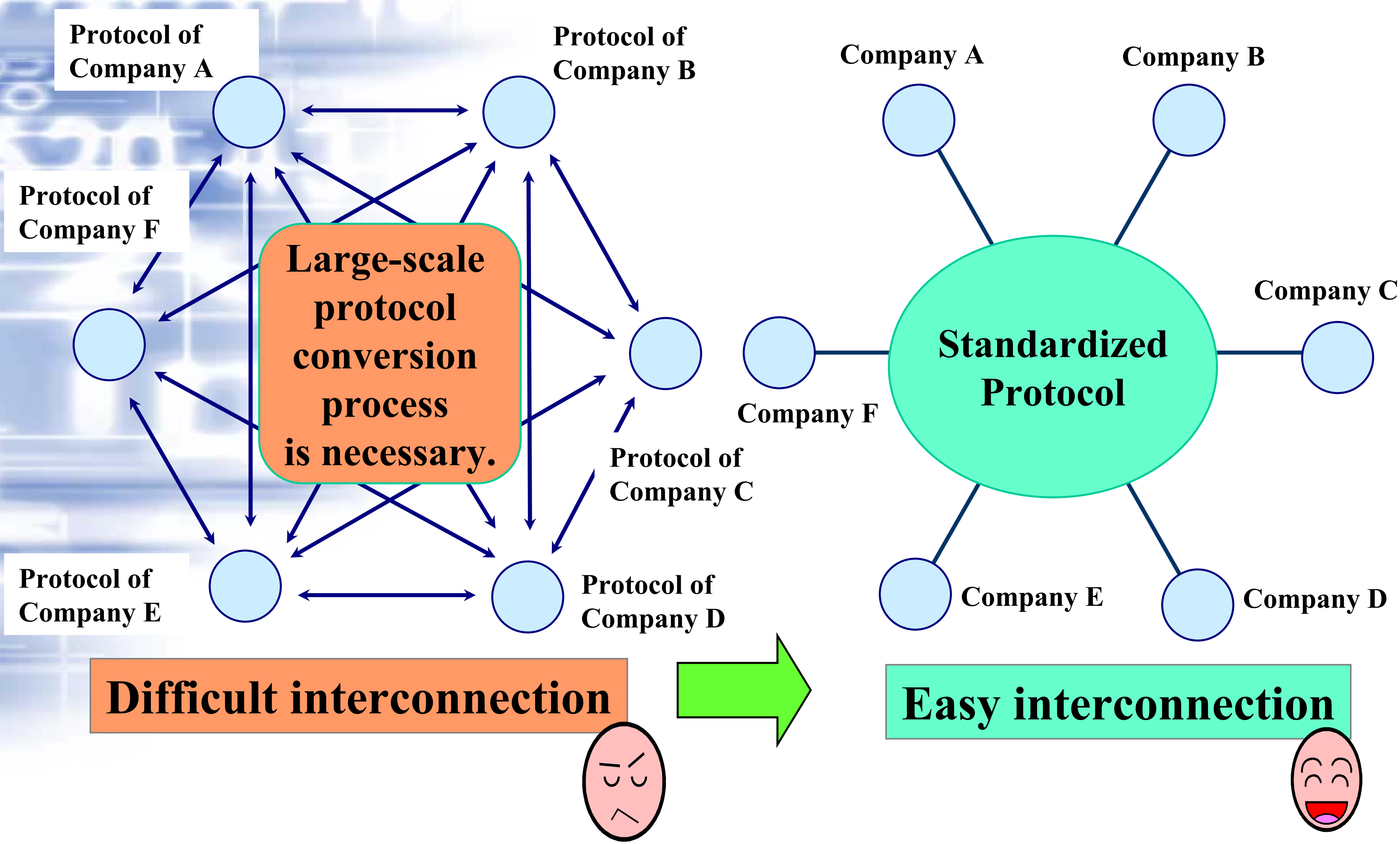
Multimedia Comm. TILC

DSL TILC*2

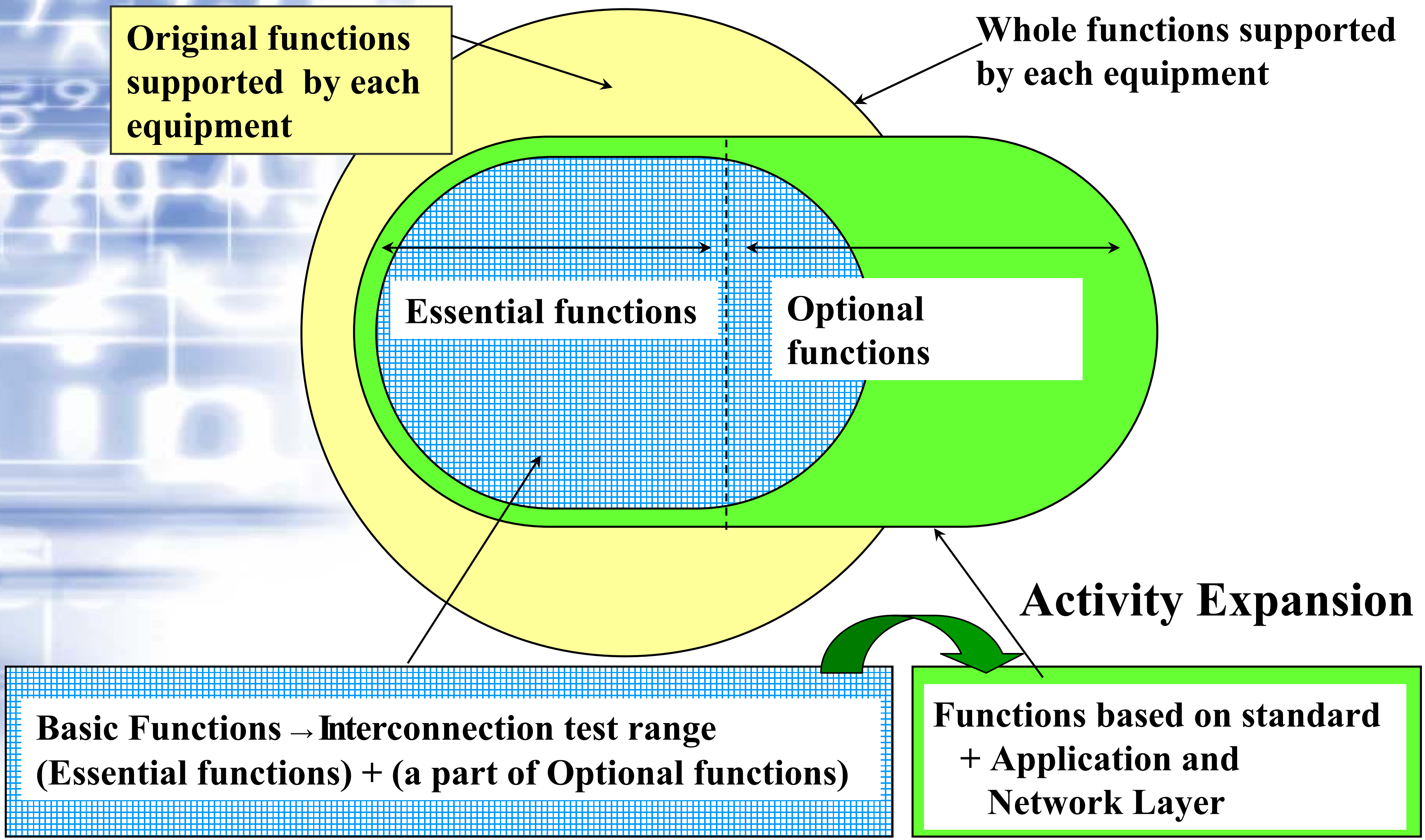
Note:
*1: Name of TILC has changed because of their activities.
*2: Those TILC has terminated on 2005.

Secretariat: Communications and Information network Association of Japan (CIAJ)

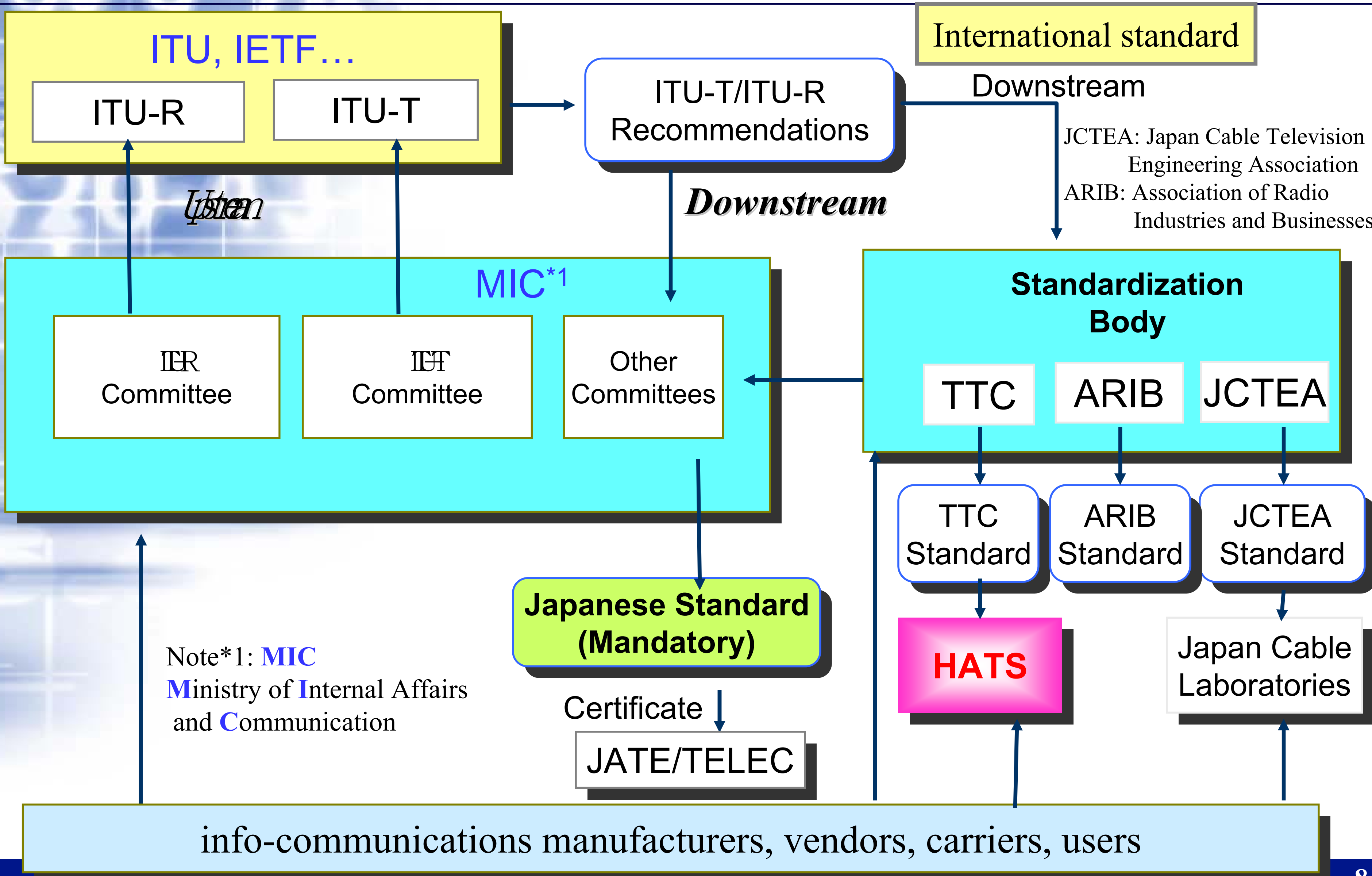
Merits of Standardization



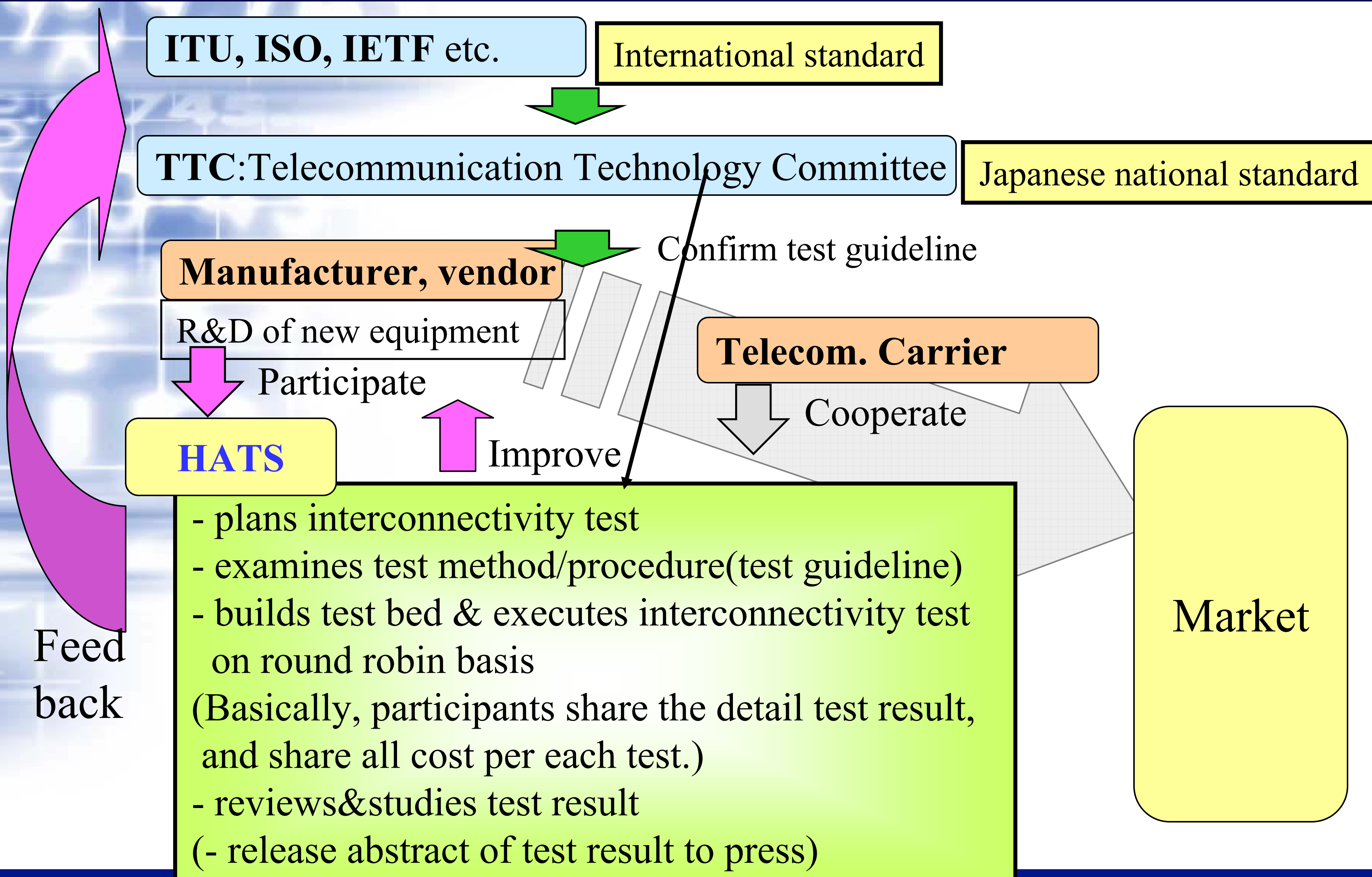
The Interconnectivity Test Range Targeted by HATS



Standardization Flow in Japan



Role of HATS



ITU, ISO, IETF etc.
International standard
TTC:Telecommunication Technology Committee
Japanese national standard
Confirm test guideline
Manufacturer, vendor
R&D of new equipment
Participate
Telecom. Carrier
Cooperate
HATS
Improve
- plans interconnectivity test
- examines test method/procedure(test guideline)
- builds test bed & executes interconnectivity test on round robin basis
(Basically, participants share the detail test result, and share all cost per each test.)
- reviews&studies test result
(- release abstract of test result to press)
Market
Feed back

Variety of HATS Test

1989- ISDN Terminal Adapter/digital telephone, G4 facsimile, PBX, MHS
1990- Analog videophone
1991- Digital videophone/H.320 videoconference, LAN router
1996- Super G3 facsimile
1997- MPEG2(H.262)
1999- LAN router(ATM, IPsec), H.324 videophone, Internet facsimile
2000- H.323 videophone(over IP), Color facsimile
2001- ADSL, LAN router(IPv6 native/tunnel mode), PBX(VoIP:IP-QSIG), SIP(VoIP), Internet-FAX
2002- ADSL(CPE), LAN router(OSPF, PPoE), SIP(VoIP), H.323+, IP-PBX(VoIP:IP-QSIG+), Internet-FAX
2003- ADSL, LAN router(VRRP), sYCC colour FAX, H.323, SIP PBX(IP-QSIG)
2004- LAN router (Internet VPN: IPsec-IKE), PBX-SIP , H.323, SIP
2005- PBX-SIP, IP-FAX, SIP, MPEG4
2006- PBX-SIP, IP-FAX, SIP, MPEG4, H.264

Actual results of HATS test

ITEMS	2002	2003	2004	2005	2006	2007
PBX	3	7	5	5	5	5
Facsimile	1	5	0	11	10	0
LAN	24	11	6			
H 323	17	13	6			
SIP	29	32	23	18	20	10
MPEG4			4	5	4	2
H 264					3	4
DSL	4					
Total	78	68	44	39	42	21

Number of Info-communication Equipments Tested Under HATS Scheme
（JFY2007, as of 2007/10) □ 21
(TOTAL：JFY1988-2007) □1,277

1989- ISDN Terminal Adapter/digital telephone, G4 facsimile, PBX, MHS
1990- Analog videophone
1991- Digital videophone/videoconference, LAN router
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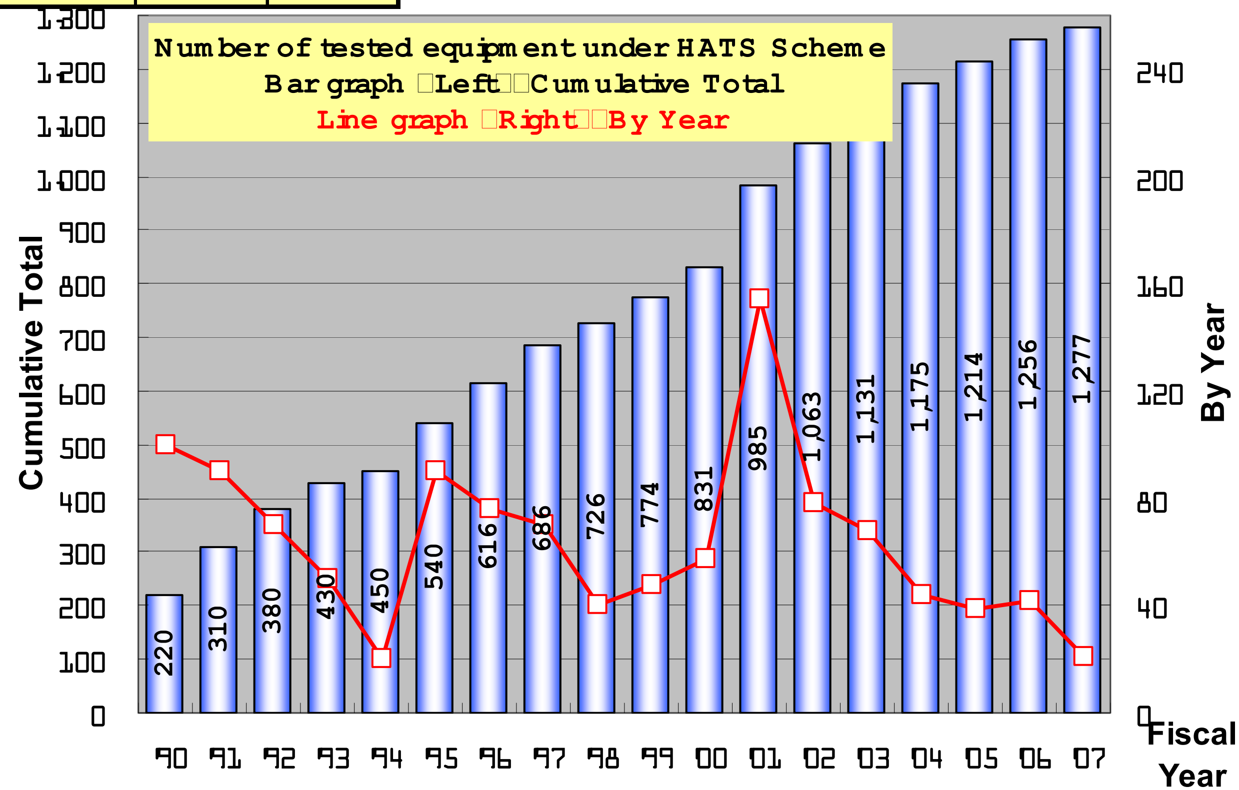
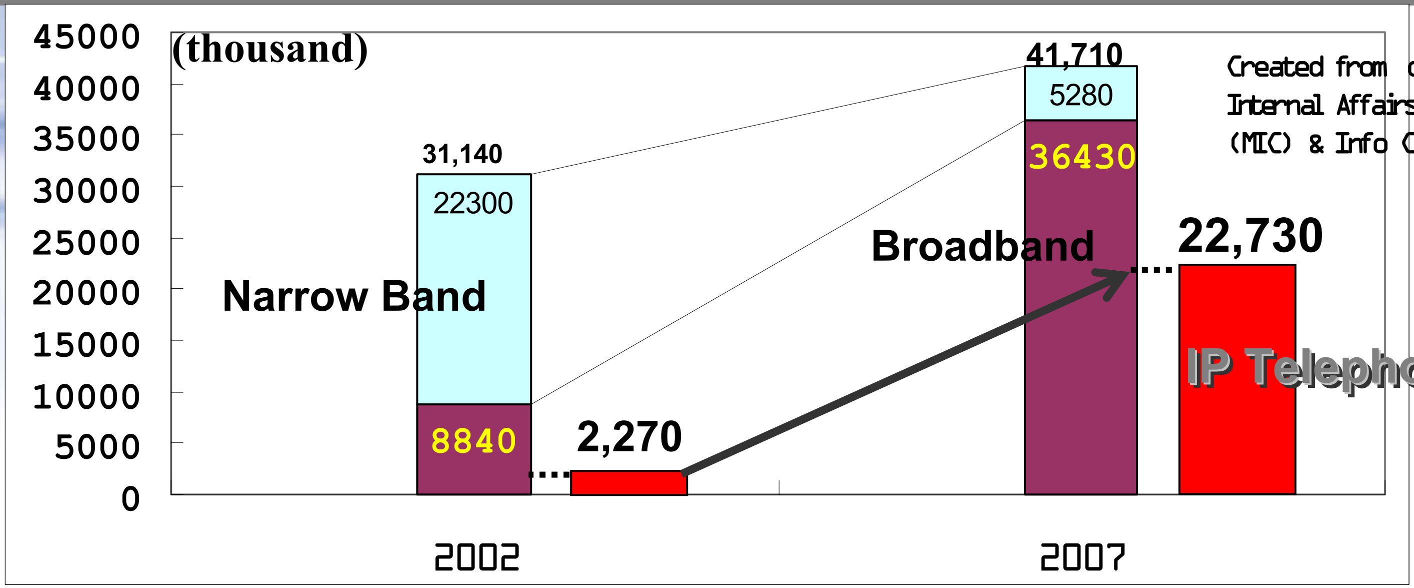


2. The trends of the IP Telephony and SIP interoperability test

Rapid growth in broadband and IP telephony services in Japan

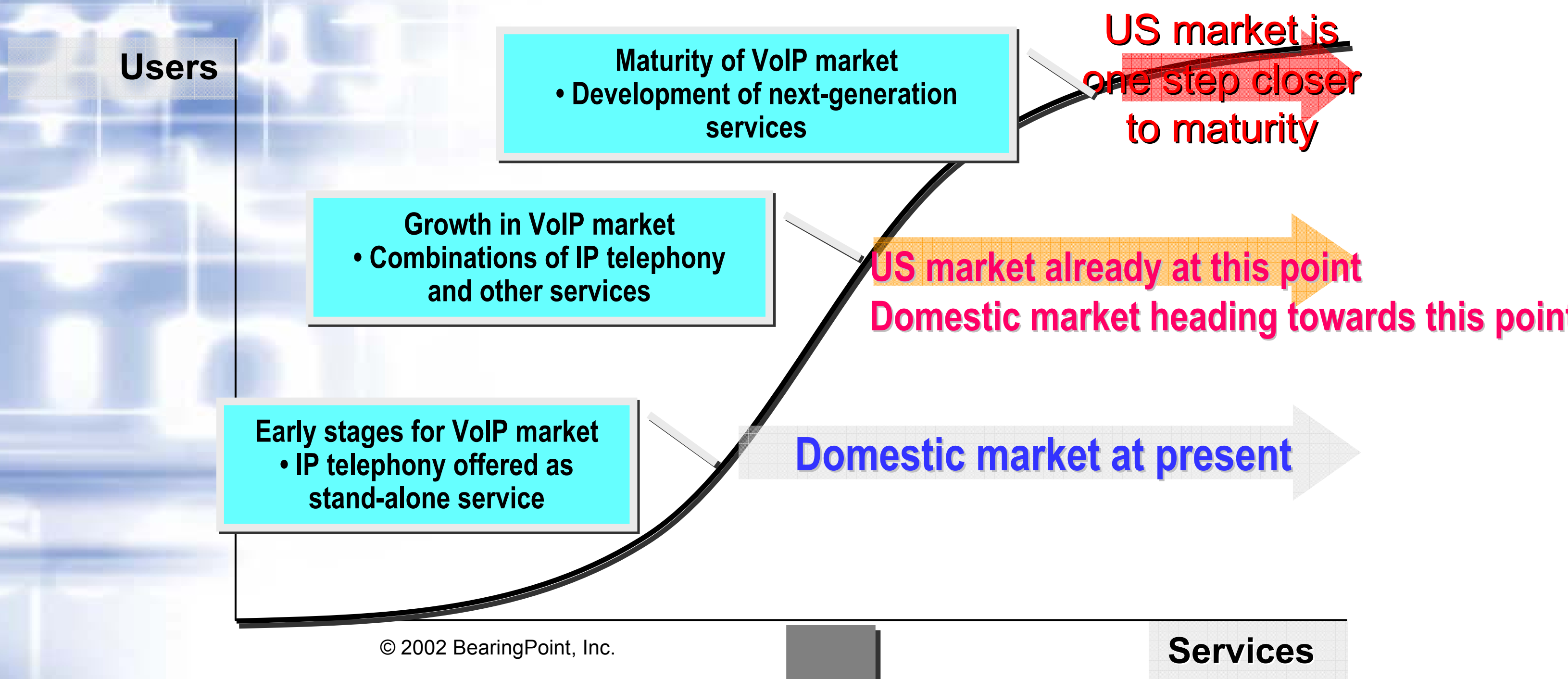
Rapid growth in the IP telephony market

- End of 2002
  - Broadband users = 8.84 million
  - IP telephony users = 2.27 million
- End of 2007
  - Broadband users = 36.43 million
  - IP telephony users = 22.73 million



Growth in the IP Telephony Market in Japan and the United States

Japan is still in the early stages of stand-alone IP telephony services, which are provided primarily by existing carriers. The future is expected to bring combined voice and data services like those seen in the United States.



In Japan, as in the United States, outsourcing demand will increase as companies seek to minimize asset risks

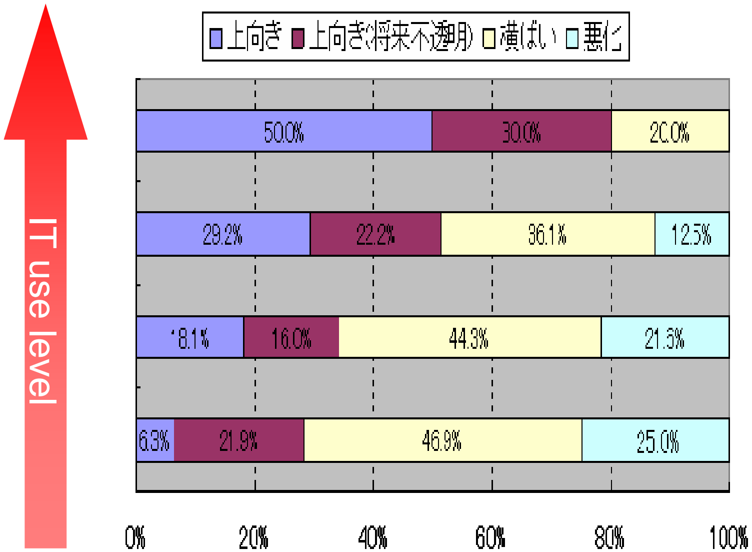
- The initial revenue base for IP telephony service operators will come from reducing corporate communication costs → emphasis on development of services
- Companies are looking to develop in-house IP Centrex facilities (including satellite capability) and housing facilities while making greater use of hosting services

Purpose for IP telephony

Not only cost reduction, but also process improvement.

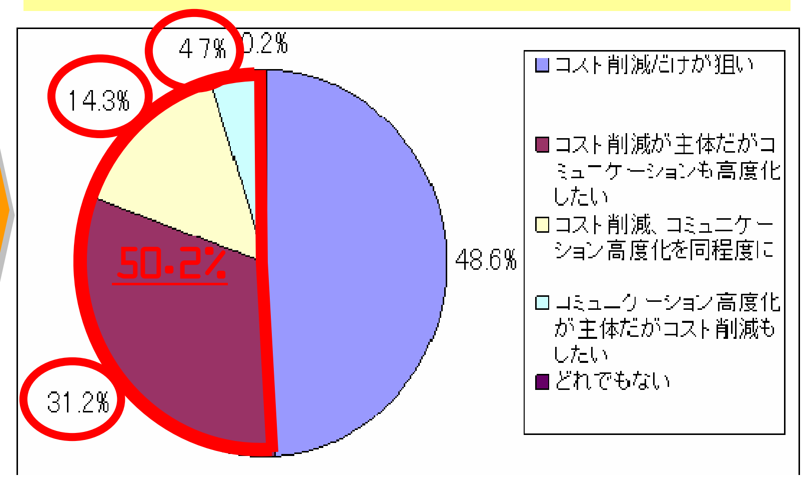
<Business condition according to IT >

The enterprise with higher IT use level tends to lead to the activation of management and the improvement of corporate performance.



< purpose of IP telephone >

In the purpose of the introduction of the IP telephone, the number of enterprises that consider using it for not only the reduction of the communication cost but also the business process improvement increases.



IP telephony → IT communication

< Past >

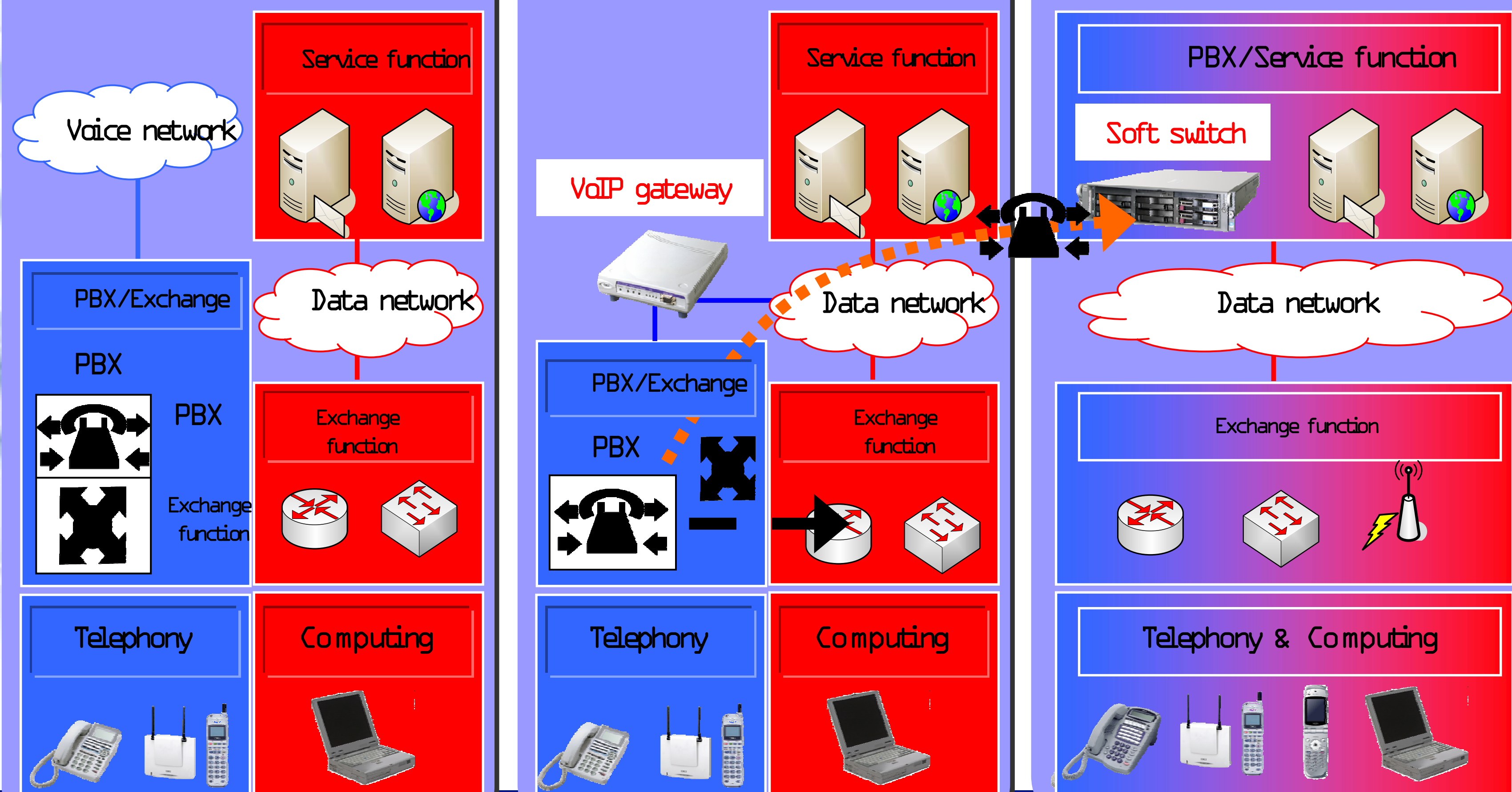
・Voice and data was on the separated network.

< IP telephony >

・Voice and data is in the same network via VoIP gateway.
(Voice communication is combined with IP-NW)

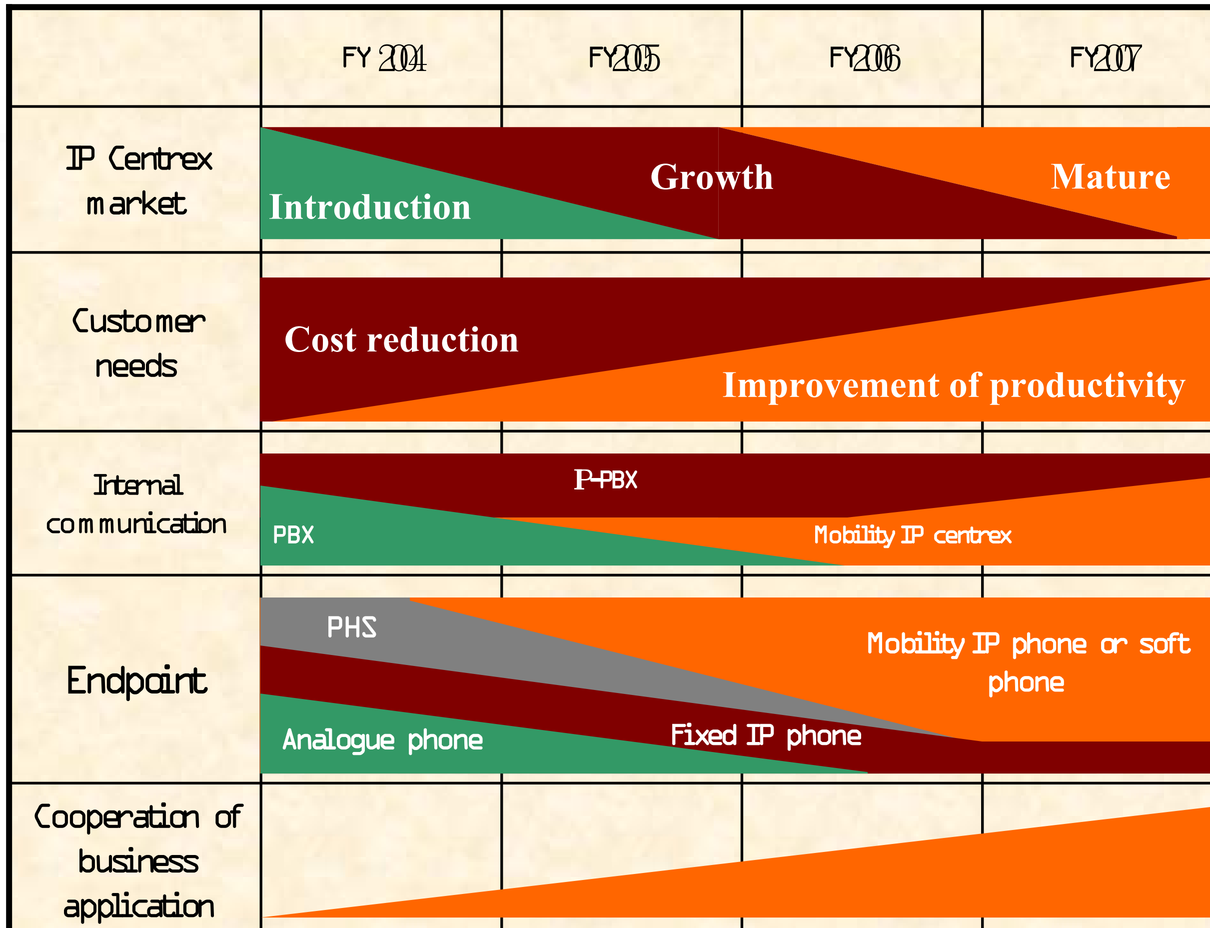
<IT communication >

・Telephone is one of IT service, and network, endpoint, and all services will be combined.



IP Centrex

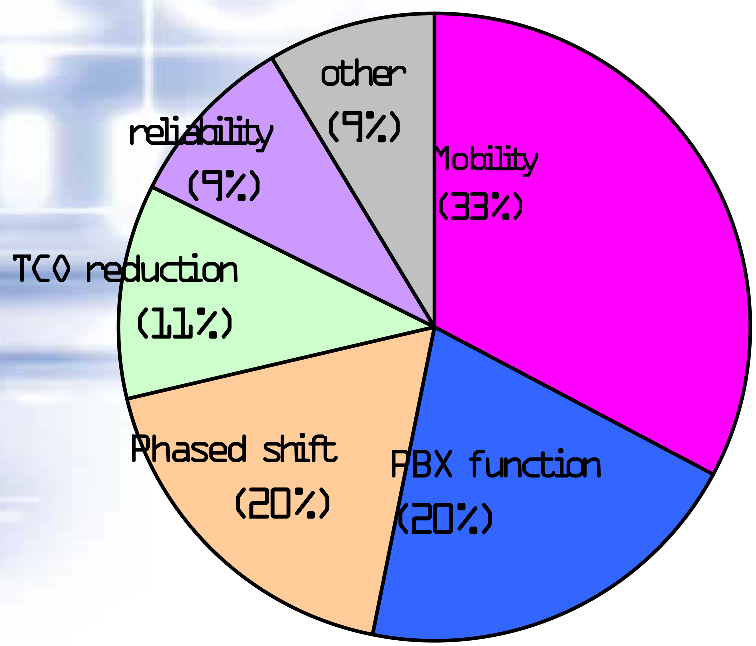
IP Centrex is not only communication cost reduction, but will bring the system optimization, and improvement of productivity!



Mobility IP centrex

Key word "Any time, and anywhere"

Most interesting thing about IP centrex?



High acknowledgment and expectation for IP telephone technology

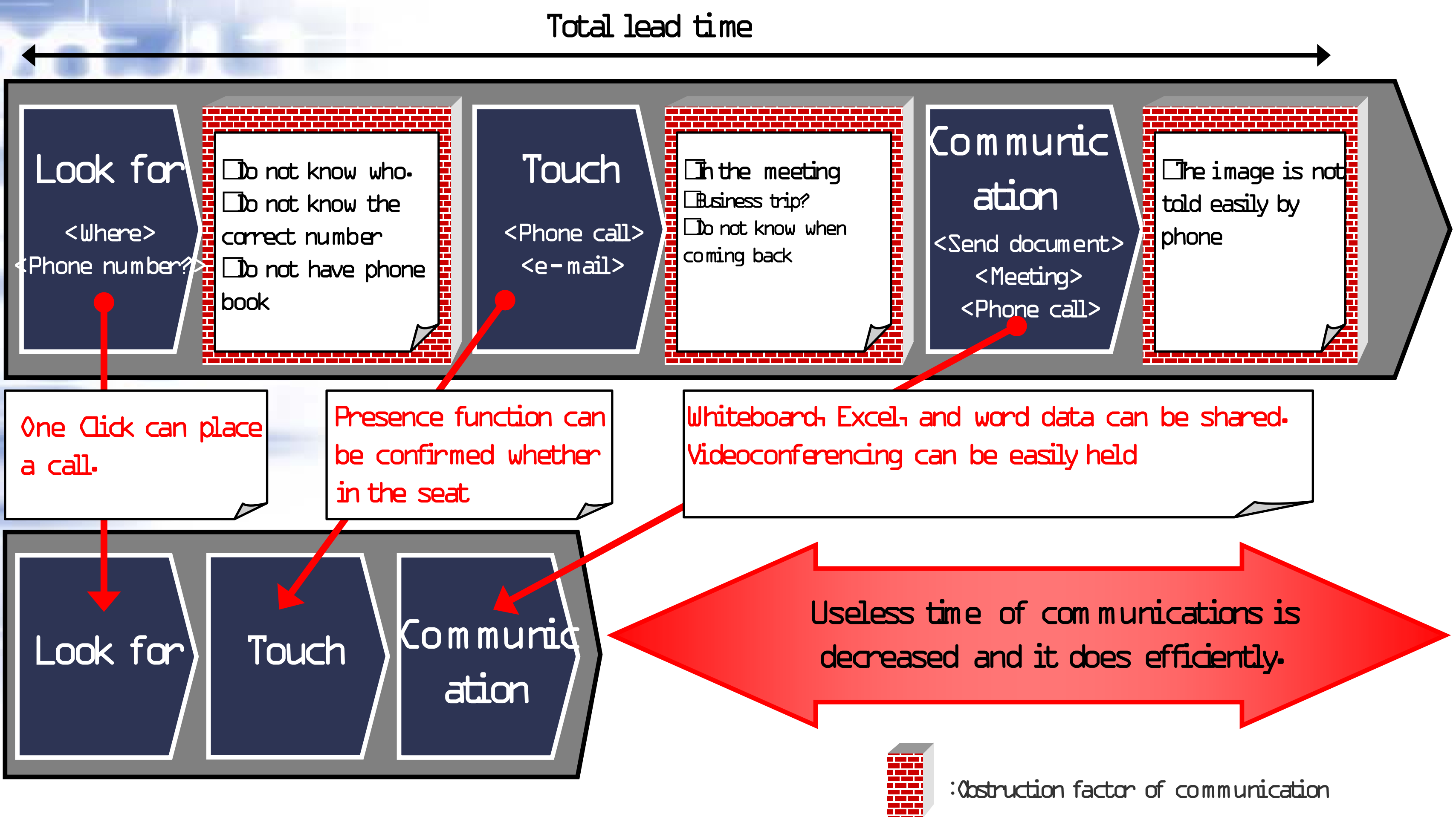
Established of work style that uses IT

Expansion of ubiquitous infrastructure (BB network, mobility LAN)

From the cost reduction to the profit expansion

Benefit that upgrade of communication system brings

The efficiency improvement of communications in the enterprise are achieved.



IP Telephony Protocols

	Description	Applications
H.323	• Standard: ITU-T (Nov 1996) • Objective: Designed for multimedia transmission (including audio and video) over packet networks • Advantages: Proven track record in corporate networks; extensive range of features	• *Multimedia transmission, video conferencing* • *IP-PBX providing conventional PBX plus additional services*
SIP	• Standard: IETF RFC3261 (Jun 2002) • Objective: Designed as protocol used to open client-server sessions* Advantages: Easy to install, fast text-based operation • Highly compatible with WWW Highly compatible with mobile applications	• *Telephone software programs with browser interface* • *Mobile telephones (*3GPP)*
H.248 (MGCP/ Megaco)	• Standard: ITU-T (Jun 2000) • IETF RFC2805/3015 • Objective: H.323 network designed as expandable architecture for building larger networks • Advantages: Facilitates monitoring and control of functionally distributed networks (call control and media control)	• *Carrier-level wide-area networks* • *IP-based public telephone services*

* Session: Logical path necessary for interaction

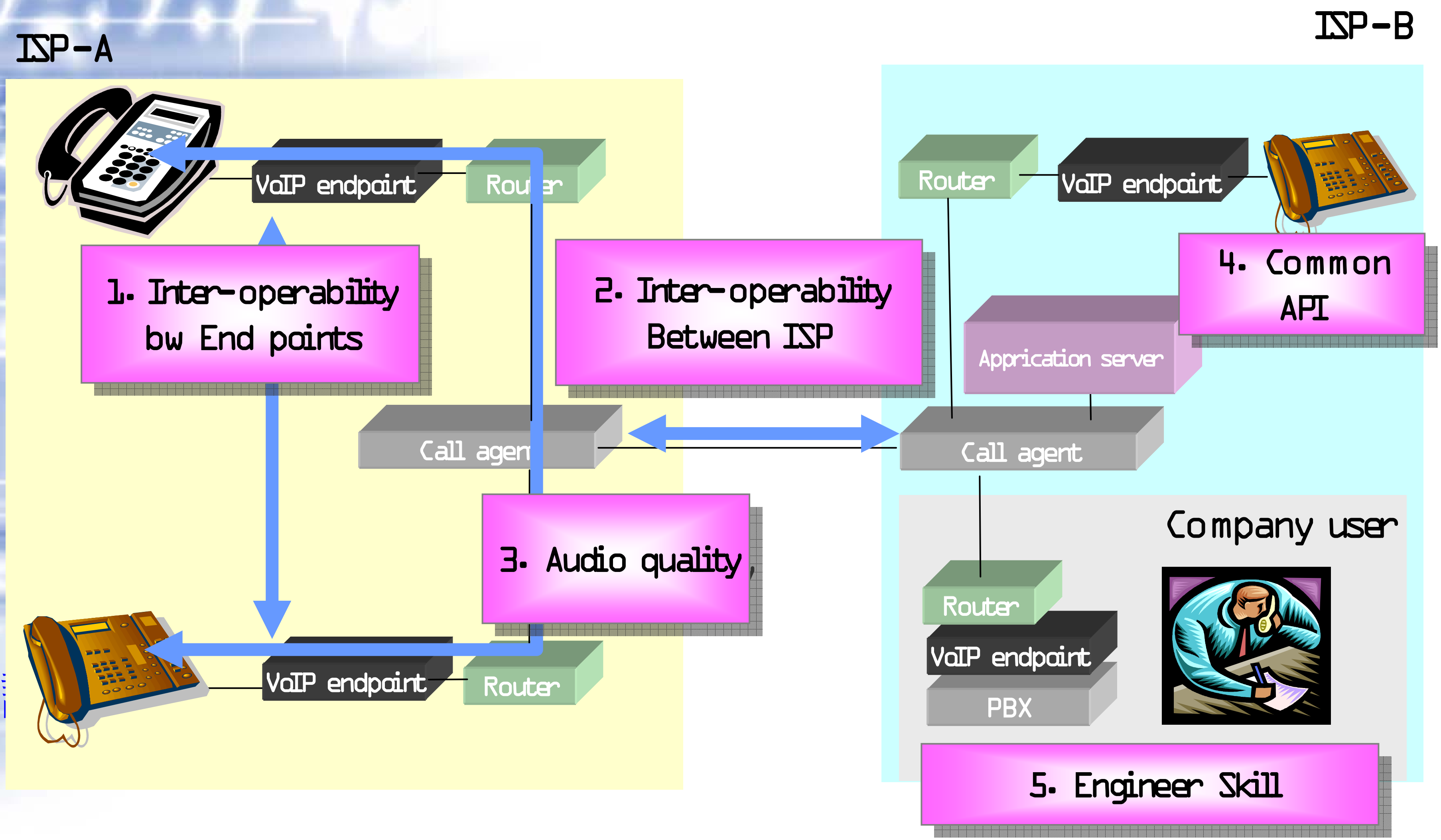
* 3GPP (3rd Generation Partnership Project): partnership project for the formulation of standards for 3G technology

What is additional service on SIP?

SIP Application

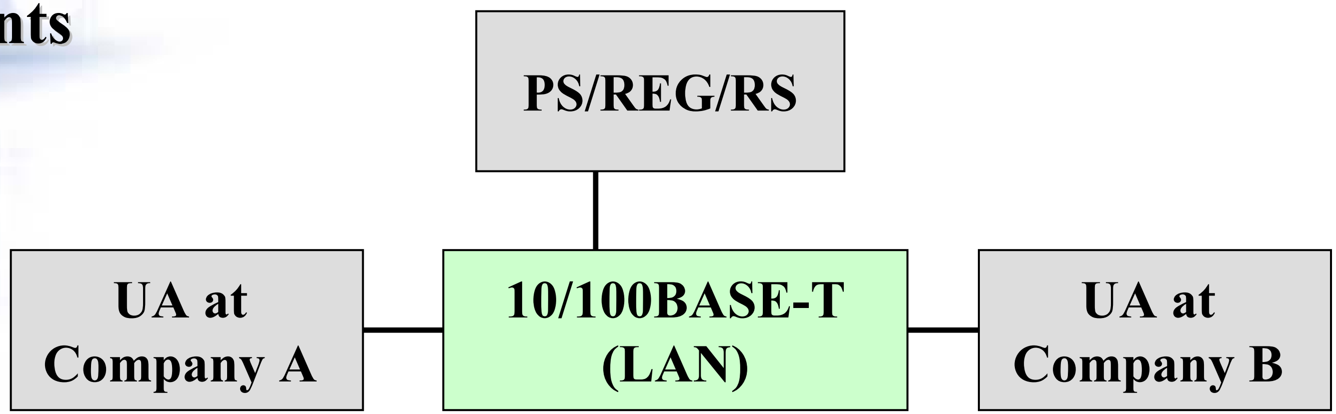
- VoIP: Voice over IP
- PoC: Push-to talk over Cellular
- IMS: IP Multimedia core network Subsystem
  PoC is one of the applications of IMS.
- Digital Household

Problems for IP telephony



Results of SIP Interconnectivity

SIP Components



- **User Agent (UA) :**

  Includes elements such as terminals and gateways which are used to establish and terminate sessions

- **Network server :**

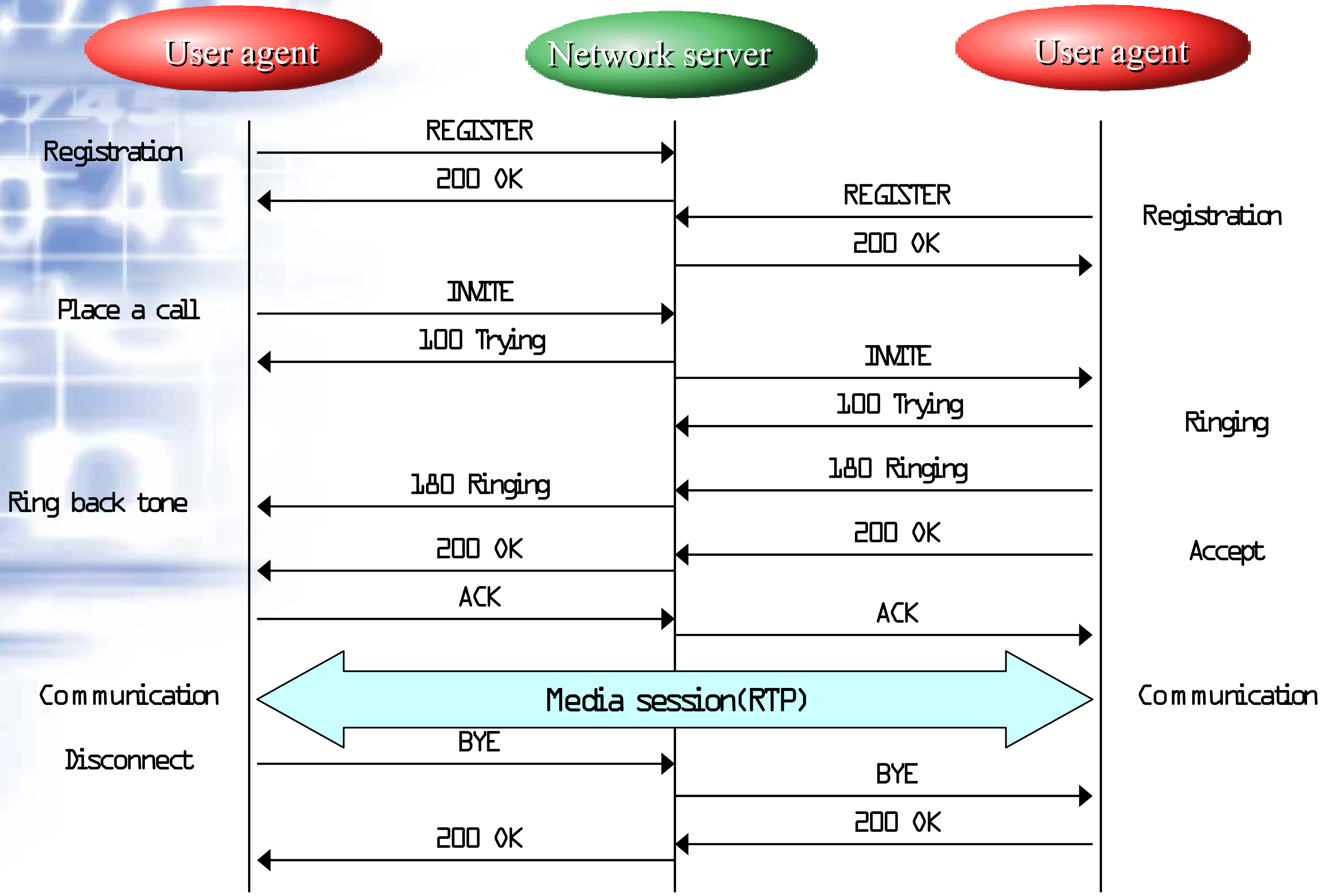
  Provides the following functionality:
  Proxy server (PS)—issues messages on behalf of other clients

  Registrar (REG)—accepts registration requests from the user agent

  Redirect server (RS)—changes the address when the user moves

  Location server—registers user position information (NB: beyond the scope of SIP)

Basic SIP call flow



CIAJ

Harmonization of Advanced Telecommunication Systems HATS

Test Procedure

- Scenario 1: UAs connected without server
- Scenario 2: UAs connected through server

- UA registered at server (Scenario 2 only)
- Transmission from UA at Company A to UA at Company B for three minutes; cut off by UA at Company A
- Transmission from UA at Company A to UA at Company B for three minutes; cut off by UA at Company B
- Transmission from UA at Company B to UA at Company A for three minutes; cut off by UA at Company B
- Transmission from UA at Company B to UA at Company A for three minutes; cut off by UA at Company A

From Guidelines for Interconnectivity Trials on Multimedia Communication Systems Using RFC3261 (SIP)—Step 1

For the details of Guide Line,
http://www.ciaj.or.jp/hats/e/activity/guideline.html

Summarized Results from Interconnectivity Trial (Check sheet)

Appendix 1 Check sheet

Test number/combination number □E□ □ □ □E1□non-P□

SIP Interconnectivity Trial—Check sheet

Date/time	
Location	

□□ □	□Company/organization:	Machine/model:	Supervisor:□□□□□ □□□
□□ □	□Company/organization:	Machine/model:	Supervisor:□ □ □□□
Server C	□Company/organization:	Machine/model:	Supervisor:□□ □□□□□□

Test items

No.		Item	Standard decision	Result (O = OK, X = NG)	Remarks (faults, problems, etc.)
1	Calling from terminal A	Audio transmission OK	Check for proper audio and video transmission in all modes. Record modes checked.		Transmission encoding mode Reception encoding mode
2		Video transmission OK	Record highest transmission speed where capacity exchange is achieved.		Transmission encoding mode Reception encoding mode
3		Video transmission speed	Should disconnect properly when disconnection is initiated by the remote side.	bps	
4		Other			
5		Remote disconnect	Should disconnect properly when disconnection is initiated by the local side.		
6		Local disconnect	Check for proper audio and video transmission in all modes. Record modes checked.		
7	Calling from terminal B	Audio transmission OK	Check for proper audio and video transmission in all modes. Record modes checked.		Transmission encoding mode Reception encoding mode
8		Video transmission OK	Record highest transmission speed where capacity exchange is achieved.		Transmission encoding mode Reception encoding mode
9		Video transmission speed	Record highest transmission speed where capacity exchange is achieved.	bps	
10		Other			
11		Remote disconnect	Should disconnect properly when disconnection is initiated by the remote side.		
12		Local disconnect	Should disconnect properly when disconnection is initiated by the local side.		

— MEMO —

Fault analysis (select one of the options below and enter in left-hand column)	
1. Fault at Company A end 2. Fault at Company B end 3. Fault at both ends 4. Connection not possible due to discrepancy between specifications	

□Test results□
□□ Result OK
×□ Result NG (see description of problem in Remarks column)
□□ Not part of test item or connection not possible due to discrepancy between specifications (see

□□ To record results from second and subsequent connection tests conducted in connection with prep time or follow up, copy this sheet using the Edit à Move or Copy Sheet function in Excel.
□□ After confirmation by both participants, the results of the interconnectivity test(s) should be forwarded to the Secretariat file server PC via an online file write procedure.

Test date/time and location
Company name, terminal, proxy information, etc.

Results for call originated at Terminal A

Results for call originated at Terminal B

Analysis of problems (if any)

Test results submitted after confirmation from both participants

Previous SIP Interconnectivity Trials

- 1st trial: February 20 –22, 2002 and April 3, 2002
  Trial involved 23 models from 14 manufacturers
- 2nd trial: July 23 – 25, 2002 and September 3, 2002
  Trial involved 28 models from 19 manufacturers
- 3rd trial: July 28 – 31, 2003 and August 26, 2003
  Trial involved 32 models from 15 manufacturers
- 4th trial: Jun 8 – 9, 2004 and July 21, 2004
  Trial involved 22 models from 16 manufacturers
- 5th trial: July 12-13, 2005 and August 25, 2005
  Trial involved 18 models from 12 manufacturers
- 6th trail :Jun 26-27, 2006 and September 3, 2006
  Trial involved 19 models from 12 manufacturers
- 7th trail :July 26, 2007
  Trial involved 10 models from 4 manufacturers

We have tested over 150 models, for the details,
http://www.ciaj.or.jp/content/pressrelease07/070808_sip.html

From next year, overseas company also join !
> We prepare English document for this trial.

Results of Connectivity Trial

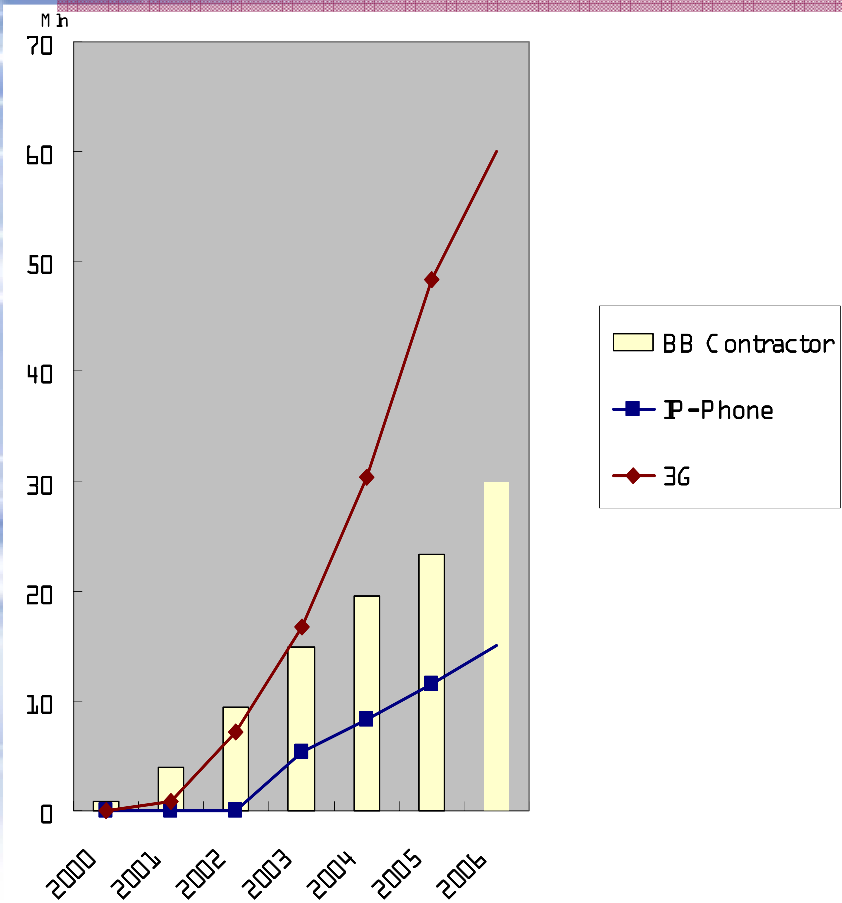
- **By last year, connections established with a success rate of 96%, but this year the rate of success rate was 100%.**
- **Every connection failures were resolved during this interoperability test**
- **For the next Step, we are planning to enhance the function and to expand the API's.**
  - **Connection test of additional telephone service function**
  - **Semi-normal examination and abnormal examination**
  - **Connection test of SIMPLE (instant message and presence)**

Problems Identified During Connectivity Trial

- Some connection failures are actually attributable to overly strict evaluation of the message body content (tag)
- The UA design verification via the server was not sufficient
- Discrepancies between versions (RFC3261/RFC2543-bis) created differences in the tag interpretation priority ranking
- Some UA can only handle specific codec, and can not accept plural description in media description header.

3. The trends of the MPEG4 and interoperability test

Status of Japanese Market:
BB line, IP-Phone, 3G terminal are spreading rapidly



Source：Ministry of Internal Affairs and Communication

■Using video content at home

YouTube Broadcast Yourself

■IP network on fixed line
KDDI announced that they will finish transition in 2007,
and also BT will shift in 2009.

NTT announced that they will provide 30 million IP network lines on fixed line in 2010.

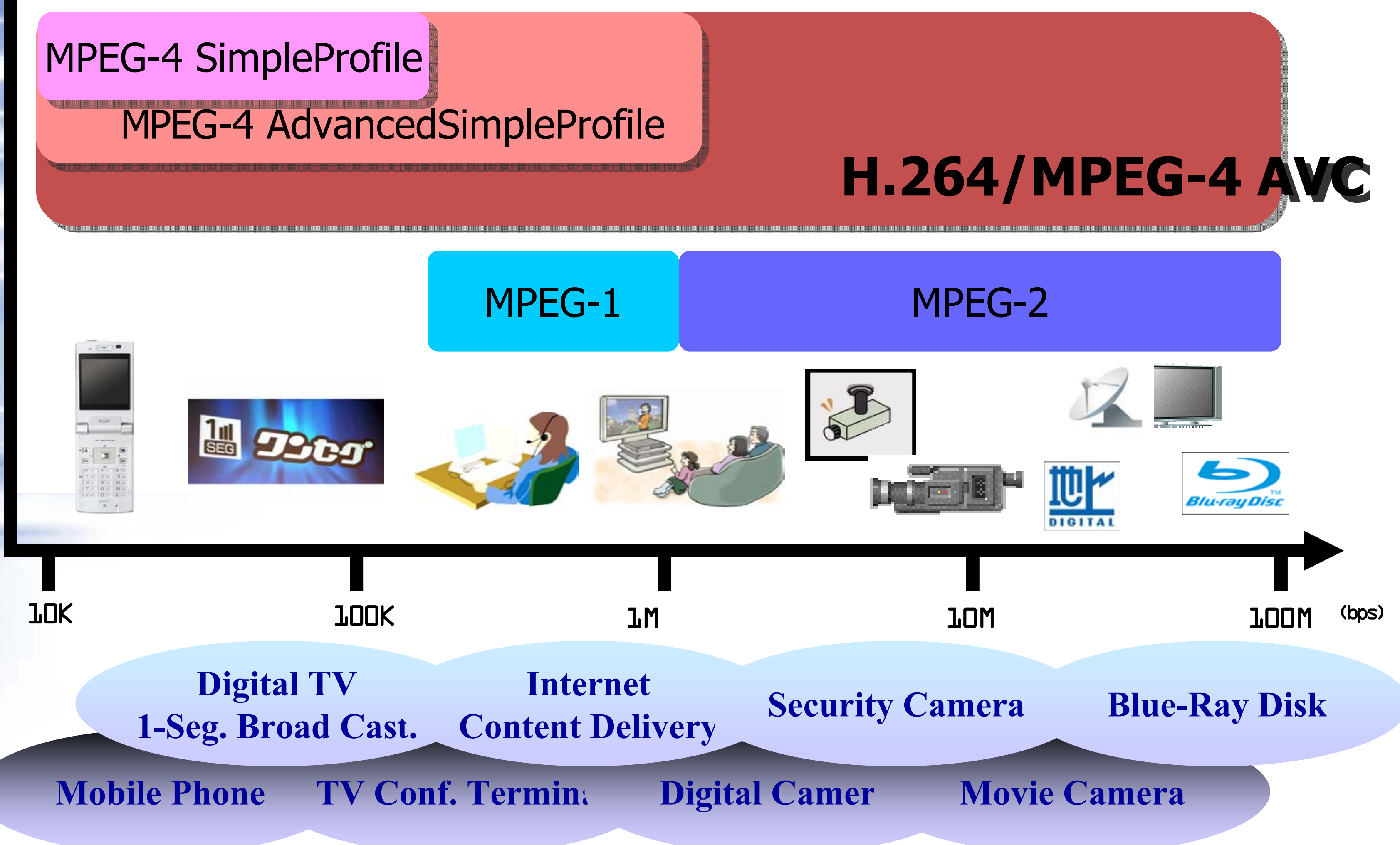
Flet's Phone VP1000

■IMS terminals are put on the market from 3 carriers in Japan now.

Push! Talk! Hello Messenger

Standard of Video codec

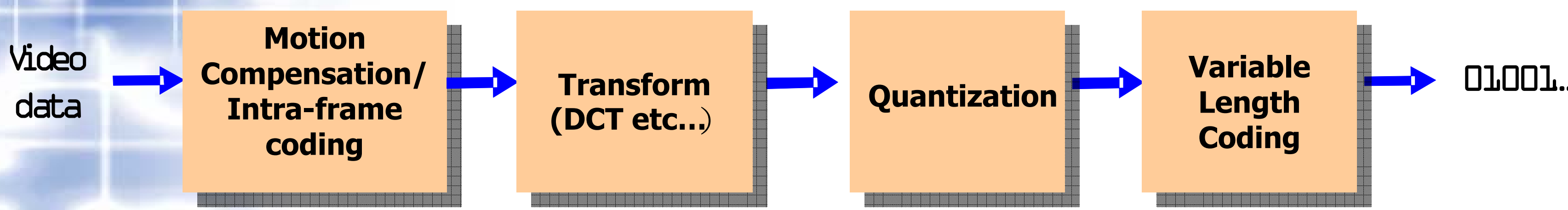
MPEG-4 is High Spec./General , Multi purpose



Outline of H.264 / MPEG-4 AVC

Latest encoding method
to improve encoding performance significantly

Enhance the existing technology with additional improvement and enable high performance compression



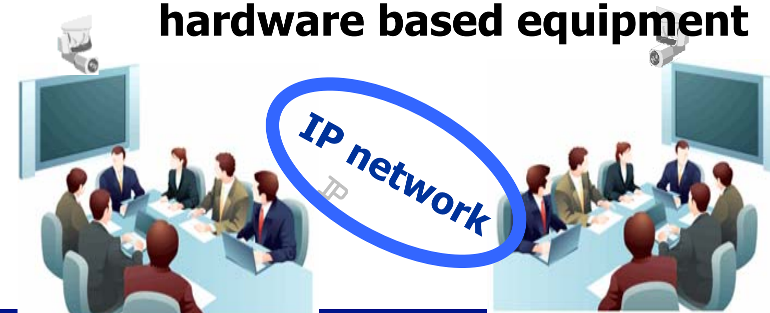
- Variable block size motion compensation
- Intra-frame coding

- Context-Adaptive Binary Arithmetic Coding Variable Length Coding (CABAC)

Compression performance = 2 times
But… Calculation amount = 10 times

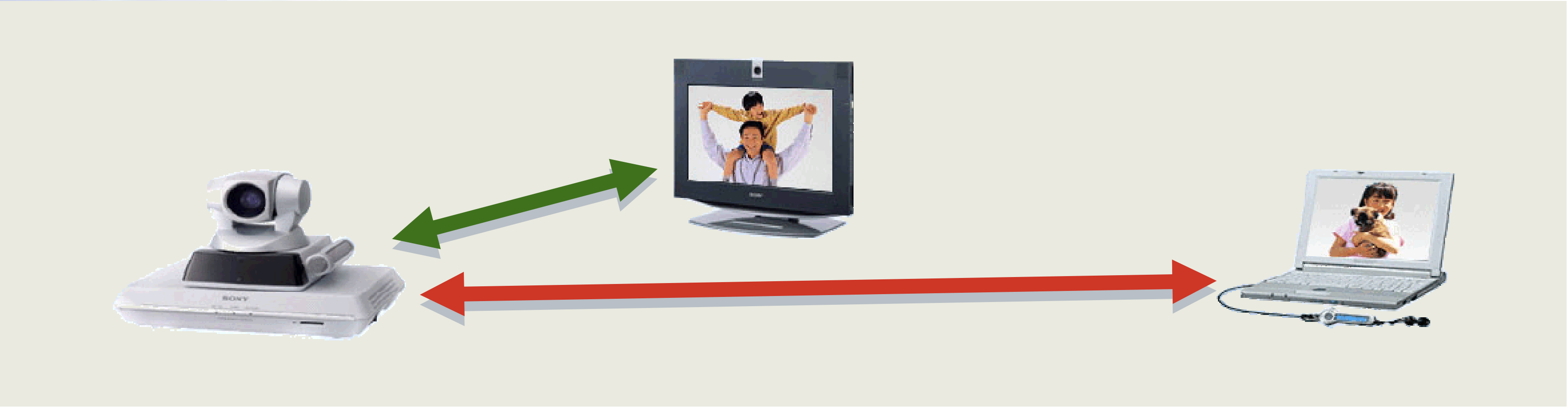
MPEG-4 for software based equipment

H.264/MPEG-4 AVC for hardware based equipment



Outline of MPEG-4 Interoperability Test

- Held 4th IOT in June, 2007
  - First IOT was held in Tokyo in 2004.
  - By now, 16 models from 16 manufacturers participated.
- Held 2nd H.264 IOT in June, 2007
  - First IOT was held in Tokyo in 2006
  - By now, 7 models from 6 manufacturers participated.
- Test under a IOT guide line
  - MPEG-4 and H.264



IOT Guideline

Added H.264 spec to the MPEG-4 guide line

■Testing Profile

Item	MPEG-4	H.264
Session Control	SIP (RFC3261), SDP (RFC2327)	
Capability Exchange	(RFC3264)	(RFC3264,**RFC3984**)
Media Transfer	RTP (RFC3550, RFC3551) , RTCP (RFC3550 Option)	
	Packatization mode(RFC3016)	Packatization mode(RFC3984)
Video (HighRate :CIF,LowRate: QCIF)	High: MPEG-4 Visual **SP@L3** Low: MPEG-4 Visual **SP@L0**	**High: H.264 (BP@L1.2)** **Low: H.264 (BP@L1)**
Audio	JT-G711 μ-Law	

■Mandatory INVITE request items (requset line, header item)

■Mandatory SDP parameters (m line : media/port, a line : profile/level)

■Make Offer/Answer model clear (using RFC3264 in SDP)

Testing Procedure

Preparation

- Confirm a specification of each vendor
- Exchange a INVITE message via ftp server

Method of testing

- Round-robin Test
- Face to Face Testing

receiver

caller

	A	B	C	D
A				
B				
C				
D				

Process

1) Call and Receive
2) Keep 3 minutes then disconnect
3) Put the result onto the test sheet
4) Exchange caller and receiver, and test above 1-3

For the details of Guide Line,
http://www.ciaj.or.jp/hats/e/activity/guideline.html

Test Sheet (H.264)

No	Item		Judging Standard	Result (Yes / No)		Remarks (problems etc…)
1	Sending side（Terminal A）	Confirmation of Audio communication	Confirm the communication Of audio and the video in each mode. Record the use mode.			Sending Side Encoding mode Receiving Side Encoding mode
2		Confirmation of Video communication				Sending side Encoding mode(Profile/Level) Receiving side Encoding mode(Profile/Level)
3		Transmission rate Of Video	Record the maximum transmission rate capability that was exchanged.	bps bps		Sending side Transmission Rate Receiving side Transmission Rate
4		RTP confirmation	Confirm the packetization mode of RFC3984.			When transmitted with Single NAL Unit, fill in Yes, otherwise, fill in No
			Confirm that the PPS/SPS is transmitted.			When transmitted, fill in Yes, When not transmitted, fill in No
5		Disconnection by A	Confirm that Terminal A disconnected properly when Terminal B disconnected.			
6		Disconnection by B	Confirm that Terminal B disconnected properly when Terminal A disconnected.			

※The problem is filled in on Memo section or remarks to the extent possible.

Result of Interoperability Test

Succeeded in all testing pairs

- MPEG-4 and H.264
  - Press released on CIAJ Home Page (8th Aug.)
    http://www.ciaj.or.jp/content/pressrelease07/070808_mpeg4.html
  - Result was Exhibited at CEATEC JAPAN 2007

- Approach from now on
  - Version up a H.264 guide line
  - Discuss optional test items
  - English Guide Line has prepared
    - Join US next Inter-operability TEST

Conclusions

HATS is activating to ensure the interconnectivity among various Info-communication equipment in multi-carrier/vendor environment of Japan.

And now…

For the NGN (Next Generation Network), full IP-Networks are planned at several nations and organizations such as ITU-T and IETF.

In Japan NGN service will start by 2010.

So, MIC in Japan is studying to establish new communication network as the NGN about following items.

- ✓Quality& Functions, Security, Interconnections, Others

➢HATS is cooperating to that studies and committees above.

Contact Point of HATS Secretariat:

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E-mail: shimizuh@ciaj.or.jp/ ogata@ciaj.or.jp