

**KX-HTS Step by Step Guide**  
**SIP Trunk**

**July 20, 2017**

Panasonic System Networks  
PBX SE team

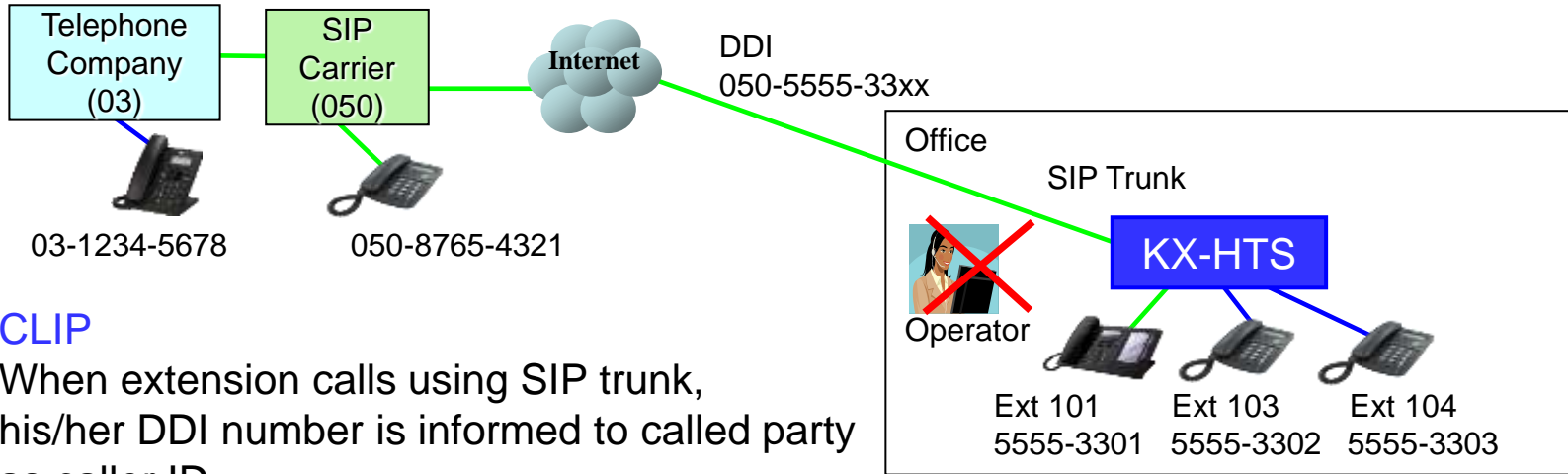
Specifications are subject to change without notice.

# 1. Overview

SIP trunk is useful in order to **save cost**.

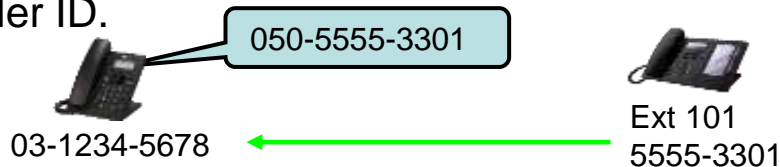
KX-HTS supports **200 DDI** numbers for one or two SIP carriers.

Extension can be called directly by DDI **without operator**.



## CLIP

When extension calls using SIP trunk, his/her DDI number is informed to called party as caller ID.



## 2. Table of Contents

Chapter	Contents
1	SIP trunk programming
2	Test after SIP trunk programming
3	ARS Programming
4	Peripheral Settings (NAT, Routing)

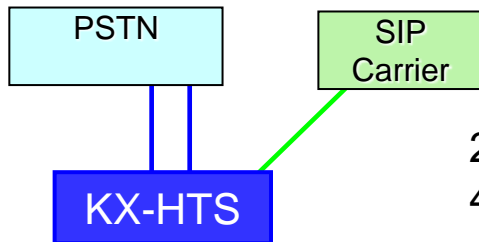
# Chapter 1

# SIP Trunk Programming

# 11. Click Port to change to SIP.

Click port 3 for example.

The screenshot shows the HTS Web Maintenance Console interface. The title bar reads "HTS Web Maintenance Console 000.00285". On the left is a "PBX Configuration" sidebar with a tree view containing: 1. System, 2. Extension, 3. Trunk, 1. Port, 2. DIL, 3. DDI, 4. Caller ID Modify & Block, 5. DISA, 6. Analogue CO Property, and 7. SIP Trunk Property. The main area is titled "PBX Configuration" and contains three rows of extension and trunk buttons. The first row shows extensions 119-124 and 101-102, with 101 and 102 highlighted in cyan. The second row shows extensions 111-118, with 116 highlighted in cyan. The third row shows extensions 103-110 and trunks 1-4, with trunk 3 highlighted by a red square.



2 lines of Analog CO and  
4 ch of SIP for example.

# 12. Change the port to SIP.

**HTS** Web Maintenance Console  
000.00285

**PBX Configuration**

- 1. System
- 2. Extension
- 3. Trunk
  - 1. Port
  - 2. DIL
  - 3. DDI
  - 4. Caller ID Modify & Block
  - 5. DISA
  - 6. Analogue CO Property
  - 7. SIP Trunk Property
- 4. TRS/ARS

**Port**

PBX Configuration > 3.Trunk > 1.Port

CO Line Number 3

Attribution

Trunk Group

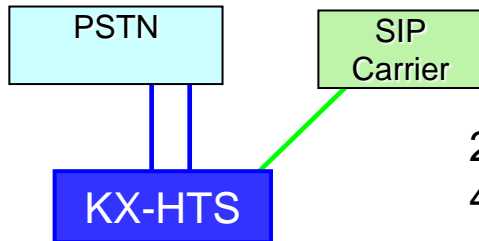
**Analogue - Basic**

Analogue

SIP Carrier - 1

SIP Carrier - 2

No Connect



2 lines of Analog CO and  
4 ch of SIP for example.

# 13. Assign Trunk Group.

**HTS** Web Maintenance Console  
000.00285

**PBX Configuration**

- 1. System
- 2. Extension
- 3. Trunk
  - 1. Port
  - 2. DIL
  - 3. DDI
  - 4. Caller ID Modify & Block
  - 5. DISA
  - 6. Analogue CO Property
  - 7. SIP Trunk Property
- 4. TRS/ARS

**Port**

PBX Configuration > 3.Trunk > 1.Port

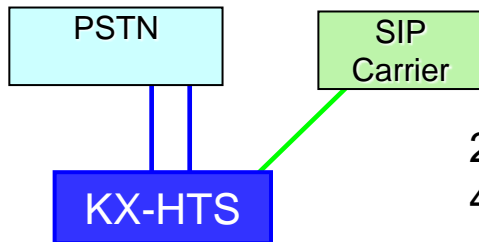
CO Line Number: 3

Attribution: SIP Carrier - 1

Trunk Group: 1

**Analogue - Basic**

1
1
2
3
4



2 lines of Analog CO for TRG1  
4 ch of SIP for **TRG2**

# 14. Repeat to other trunk.

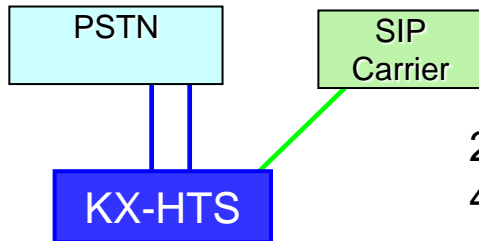
**HTS** Web Maintenance Console  
000.00285

**PBX Configuration**

- 1. System
- 2. Extension
- 3. Trunk
  - 1. Port
  - 2. DIL
  - 3. DDI
  - 4. Caller ID Modify & Block
  - 5. DISA
  - 6. Analogue CO Property
  - 7. SIP Trunk Property

**PBX Configuration**

Extension	119	120	121	122	123	124	101	102	Doorphone	1	2		
	17	18	19	20	21	22	23	24					
Extension	111	112	113	114	115	116	117	118	Trunk	5	6	7	8
	9	10	11	12	13	14	15	16					
Extension	103	104	105	106	107	108	109	110	Trunk	1	2	3	4
	1	2	3	4	5	6	7	8					



2 lines of Analog CO and  
4 ch of SIP for example.



# 15. Assign SIP account.

Ask your account to your SIP carrier.

**HTS** Web Maintenance Console  
000.00285

**PBX Configuration**

- 1. System
- 2. Extension
- 3. Trunk
  - 1. Port
  - 2. DIL
  - 3. DDI
  - 4. Caller ID Modify & Block
  - 5. DISA
  - 6. Analogue CO Property
  - 7. SIP Trunk Property
- 4. TRS/ARS
- 5. System Speed Dialling
- 6. Conference
- 7. Voice Mail

**Network Configuration**

**Maintenance**

### SIP Trunk Property

PBX Configuration > 3. Trunk > 7. SIP Trunk Property

**Common** **SIP Carrier - 1** SIP Carrier - 2

SIP Registration Status: Not yet

Provider Name (20 characters):

SIP Server Name (100 characters):

SIP Server IP Address:  0  . 0  . 0  . 0

SIP Server Port Number:  5060 ( 1 - 65535 )

SIP Service Domain (100 characters):  [ 0-9 a-z A-Z - . ]

Subscriber Number:  [ 0-9 # \* ]

**Account**

User Name (64 characters):

Authentication ID (64 characters):

Authentication Password (32 characters):

**Register**

Register Ability:  Enable  Disable

# 16. Assign Voice Codec.

Ask correct CODEC to your SIP carrier.

If it is G.729, assign G729 for 1<sup>st</sup> priority and None for 2<sup>nd</sup> and 3<sup>rd</sup> priority.

PBX Configuration > 3.Trunk > 7.SIP Trunk Property

## Voice

- |                       |  |
|-----------------------|--|
| IP Codec Priority 1st | <input checked="" type="radio"/> G.711A<br><input type="radio"/> G.711Mu<br><input type="radio"/> G.729A                               |
| IP Codec Priority 2nd | <input type="radio"/> G.711A<br><input checked="" type="radio"/> G.711Mu<br><input type="radio"/> G.729A<br><input type="radio"/> None |
| IP Codec Priority 3rd | <input type="radio"/> G.711A<br><input type="radio"/> G.711Mu<br><input type="radio"/> G.729A<br><input checked="" type="radio"/> None |

# 18. Assign DDI.

**HTS** Web Maintenance Console  
000.00285

**PBX Configuration**

- 1. System
- 2. Extension
- 3. Trunk
  - 1. Port
  - 2. DIL
  - 3. DDI
  - 4. Caller ID Modify & Block
  - 5. DISA
  - 6. Analogue CO Property
  - 7. SIP Trunk Property
- 4. TRS/ARS
- 5. System Speed Dialling
- 6. Conference
- 7. Voice Mail

**Network Configuration**

**Maintenance**

**DDI**  
PBX Configuration > 3.Trunk > 3.DDI

**Dialling Plan**

SIP Carrier	DDI	Remove Digit	Additional Dial [ 0-9 # * ]
1	Enable	0	
2	Enable	0	

**DDI Table**

No.	DDI Number [ 0-9 # * ]	Name	Destination Day	Destination Lunch	Destination Night
1	55553301		101/	111/	501/DISA
2	55553302		102/	602/Group	602/Group
3	55553399		691/Meet Me	691/Meet Me	691/Meet Me
4					

When received DDI number after modification by “Remove” and “Add” cannot be found in DDI table, but it is same as extension number, destination becomes the extension. If it is not extension number also, DIL is applied.

# 19. Assign CLIP.

**HTS** Web Maintenance Console  
000.00285

**PBX Configuration**

- 1. System
- 2. Extension
- 1. Port
- 2. Phone
- 3. Flexible Buttons
- 4. Extension Group

**Port**  
PBX Configuration > 2.Extension > 1.Port

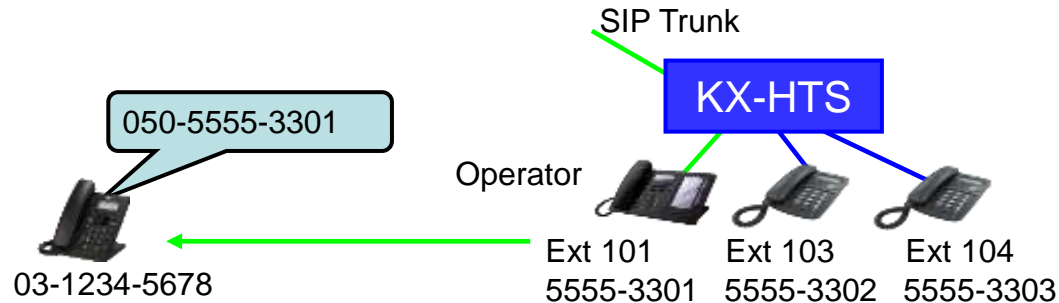
Extension Number:  [ 0-9 ]

Extension Name:

**CLIP**

SIP Carrier - 1:  [ 0-9 ]

SIP Carrier - 2:  [ 0-9 ]



# Chapter 2

## Test after SIP Programming

# 21. Confirm CO Access Number.

HTS Web Maintenance Console  
000.00285

PBX Configuration

- 1. System
  - 1. Date & Time
  - 2. MOH
  - 3. Week Table
  - 4. Numbering Plan
  - 5. Timers
  - 6. System Options
  - 7. SMDR
- 2. Extension
- 3. Trunk
- 4. TRS/ARS
- 5. System Speed Dialling
- 6. Conference
- 7. Voice Mail

Network Configuration

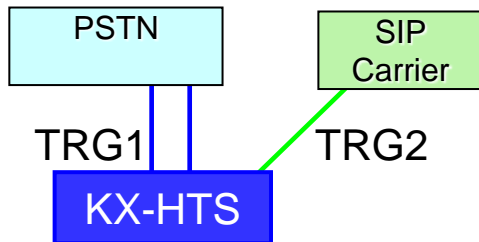
Maintenance

## Numbering Plan

PBX Configuration > 1. System > 4. Numbering Plan

Main Features Quick Dial Dialling Plan

No.	Feature	Dial (1 digit) [ 0-9 ]	Additional Dial
5	Extension Numbering Scheme 5	5	XX
6	Extension Numbering Scheme 6	6	XX
7	Call Park	7	XX
8	Trunk Line / Trunk Group	8	0X or #X
9	Idle Line Access (Local Access) - 1	9	
10	Idle Line Access (Local Access) - 2	0	
11	Feature		



Dial 0 or 9 for any trunk + Phone number.  
Dial 8#1 for TRG1 + Phone number.  
Dial 8#2 for TRG2 + Phone number.

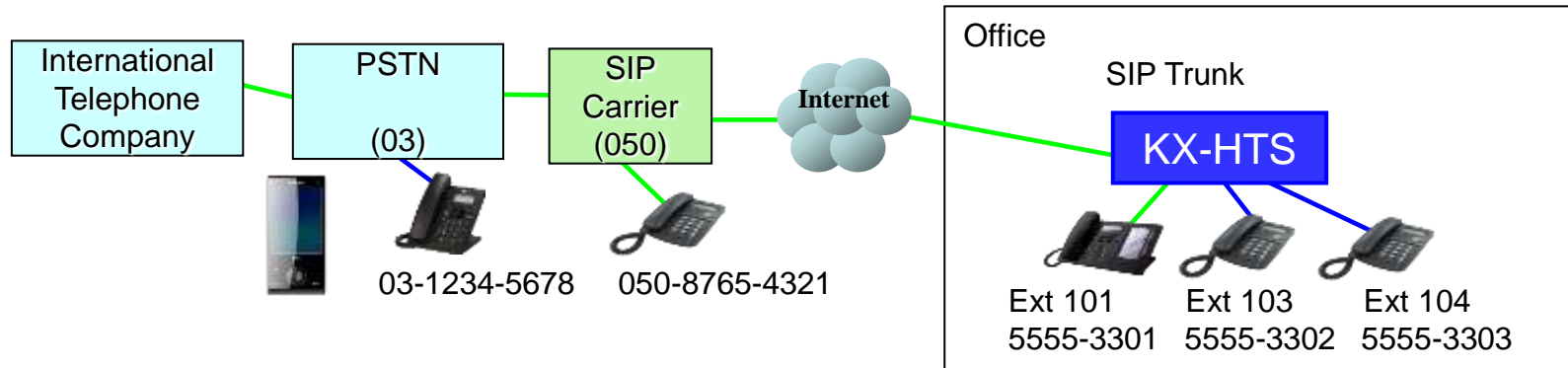
Phone number is 03-1234-5678 for example.

# 22. Make and Receive a call actually.

Outgoing call : CLIP as caller ID has to be sent to called party correctly.

Incoming call : Call has to be distributed to DDI destination with caller ID correctly.

Talking : Voice quality has to be good. DTMF can be sent.



Call	Phone	Check
Call from	Phone in SIP Carrier	
Call to	Phone in SIP Carrier	
Call from	Phone in PSTN (Analog)	
Call to	Phone in PSTN (Analog)	

Call	Phone	Check
Call from	Cellular Phone	
Call to	Cellular Phone	
Call from	International (if customer needs)	
Call to		

# Chapter 3

## ARS Programming



# 30. ARS : Automatic Route Selection

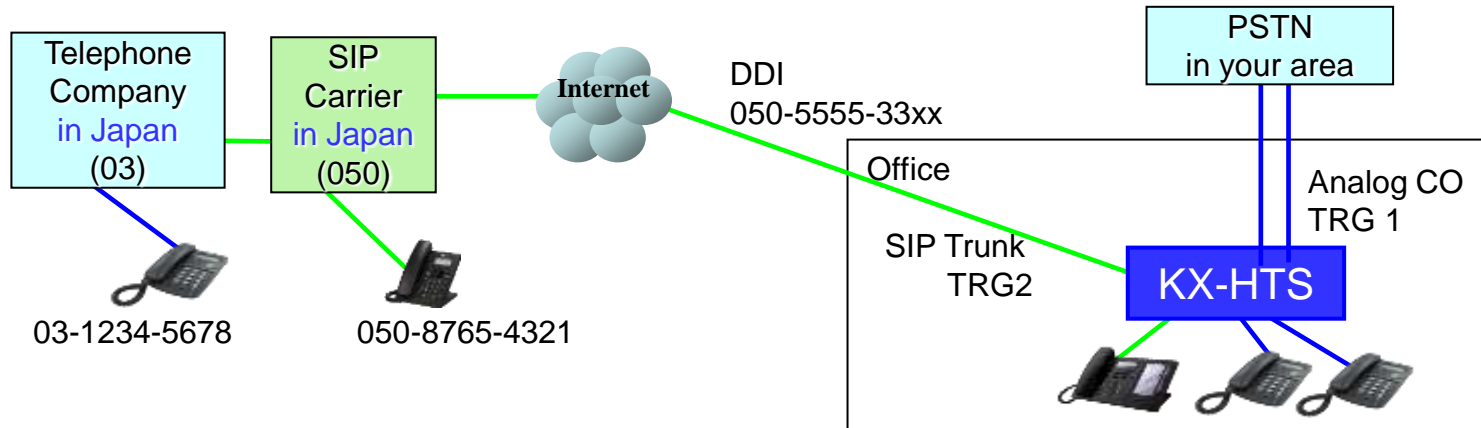
**SIP trunk** can be selected automatically for programmed phone number. Dialed number can be modified automatically.

9/0-011-81(Japan) selects SIP : TRG2. Remove 01181 and add 0.

Dialed number by user : 9/0-011-81-3-1234-5678

Dialed number by KX-HTS to SIP : 03-1234-5678

9/0-Other dial selects Analog CO : TRG1.



# 31. Assign Leading Digits & Change TRS.

**HTS** Web Maintenance Console  
000.00314

**PBX Configuration**

- 1. System
- 2. Extension
- 3. Trunk
- 4. TRS/ARS
  - 1. Leading Digits
  - 2. ARS Carrier
  - 3. Account Code
  - 4. Emergency Dial
  - 5. Options
- 5. System Speed Dialling
- 6. Conference
- 7. Voice Mail

**Leading Digits**  
PBX Configuration > 4 TRS/ARS > 1. Leading Digits

No.	Leading Digits [ 0-9 # * N X Z ]	TRS Level (COS) 1	TRS Level (COS) 2
1	011	Allow	Deny
2	01181	Allow	Allow
3	0X	Allow	Allow
4		Allow	Deny

Default of TRS level is 2 for all extensions.

# 32. Assign ARS Carrier.

No.	Leading Digits [ 0-9 # * N X Z ]	TRS Level (COS) 1	TRS Level (COS) 2	ARS Carrier Priority-1	ARS Carrier Priority-2	ARS Carrier Priority-3
1	011	Allow	Deny	1:	None	None
2	01181	Allow	Allow	2:	1:	None
3	0X	Allow	Allow	1:	None	None
4		Allow	Deny	None	None	None

If carrier name is assigned as next page, it is displayed for selection.

ARS Carrier Priority-1	ARS Carrier Priority-2	ARS Carrier Priority-3
1:Local PSTN	None	None
2:Japan SIP	1:Local PSTN	None
1:Local PSTN	None	None

# 33. Assign ARS Carrier.

Remove 5 digits (01181) and add "0".

**HTS** Web Maintenance Console  
000.00314

**PBX Configuration**

- 1. System
- 2. Extension
- 3. Trunk
- 4. TRS/ARS
  - 1. Leading Digits
  - 2. ARS Carrier
  - 3. Account Code
  - 4. Emergency Dial
  - 5. Options
- 5. System Speed Dialling
- 6. Conference
- 7. Voice Mail

**ARS Carrier**  
PBX Configuration > 4 TRS/ARS > 2. ARS Carrier

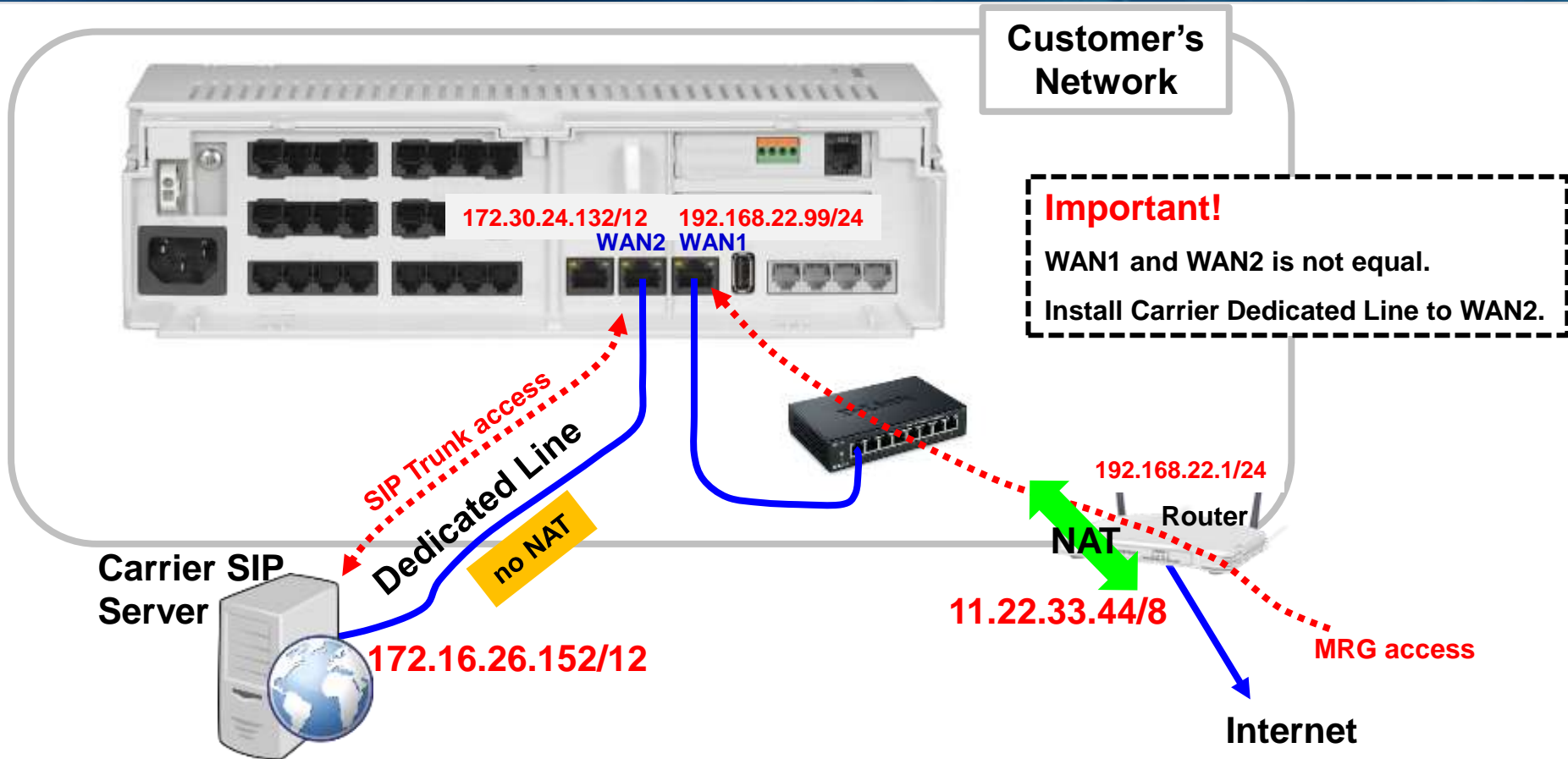
No.	Carrier Name	Dial Modification Remove(digits)	Dial Modification Add [ 0-9 # * ]	Trunk Group
1	Local PSTN	0		TRG1
2	Japan SIP	5	0	TRG2
3		0		TRG1
4		0		TRG1

Programming is completed. Try making call by ARS.

# Chapter 4

## Peripheral settings (NAT, Routing)

# 41. Diagram

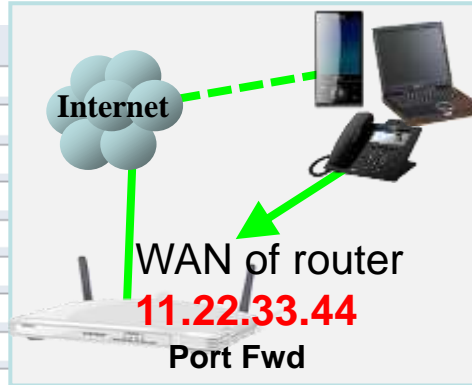


# 42. NAT Traversal (1)

## PBX Configuration > 2.Extension > 7.SIP Extension Property

### SIP Extension & SIP Trunk **Common**

Voice (RTP) UDP Port No. (Server)	12000
SIP Port Number	5060
Jitter Buffer Type for Voice	Adaptive
Jitter Buffer Delay Min. for Voice	20
Jitter Buffer Delay Max. for Voice	180
Jitter Buffer Delay Init. for Voice	50
Jitter Buffer Type for Data	Fixed
Jitter Buffer Delay Min. for Data	20
Jitter Buffer Delay Max. for Data	180
Jitter Buffer Delay Init. for Data	50
NAT Traversal	<input type="radio"/> Off <input checked="" type="radio"/> Fixed IP Address <input type="radio"/> STUN <input type="radio"/> DDNS
NAT - Fixed Global IP Address	11 . 22 . 33 . 44



Basically...

NAT Traversal works for both of

- MRG access
- SIP Trunk access

### IP Address of SIP Extension in existing router

IP Address	0 . 0 . 0 . 0
Subnet Mask	0 . 0 . 0 . 0

When set all 0(Zero), **NAT Traversal does not work for local IP Address.** HTS judge IP Address automatically.



**172.16.26.152** = Local IP Address NAT Traversal does not work!



# 42. NAT Traversal (2)

## HTS Local Address Judgement

RFC1918: Private IP Address (Local IP Address)

Class	Range
A	10.0.0.0 – 10.255.255.255/8
B	172.16.0.0 – 172.31.255.255/12
C	192.168.0.0 – 192.168.255.255/16

= Local IP Address



**No NAT Traversal**



# 43. Routing

## Network Configuration > 5.Route > 1.Static Routing

### Static Routing

The static routing function determines the path that data follows over your network before and after it passes through your router. You can use static routing to allow different IP domain users to access the Internet through this device. The default route cannot be added from this web page.

Destination IP	Subnet Mask	Gateway	Interface	
<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="button" value="Add"/>
172.16.0.0	255.255.240.0	172.30.0.1	WAN2_DHCP_Client	<input type="button" value="Delete"/>
				<input type="button" value="Cancel"/>

Destination Subnet address.    Destination Subnet netmask  
Ex: 12bits = 255.255.240.0    Default GW of network which  
WAN2 belongs.

➔ The packet heading to Carrier SIP Server **172.16.26.152** will be sent out from WAN2 port.

---

## Network Configuration > 3.WAN > 1.Setting

### WAN Setting

	WAN Name	WAN Channel	Type	Default WAN
<input type="radio"/>	WAN1_Fix_IP	WAN1	Fix IP	<input checked="" type="radio"/>
<input type="radio"/>	WAN2_DHCP_Client	WAN2	DHCP Client	<input type="radio"/>

➔ Other packet will be sent out from WAN1 (Default WAN)

**Thank you !**

# Revision

Date	No.	Change
July 28, 2016	All	First official release
August 8, 2016	Chapter 3	ARS was added.
Sept 16, 2016	12 (Page 6)	Picture was added.
	33 (Page 21)	Explanation was added.
June 28,2017	Chapter 4	Peripheral Settings(NAT, Routing) was added.
July 20,2017	17(Page11)	Side 17 "SIP Domain Name" is deeted.