KX-HTS Step by Step Guide

SIP Trunk

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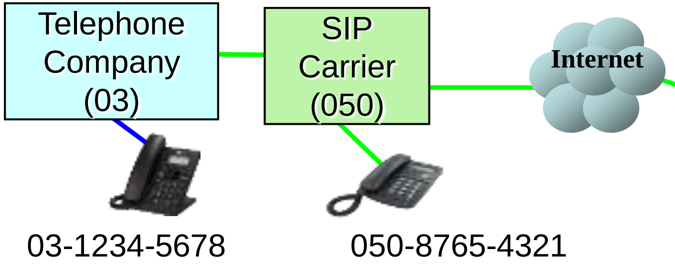
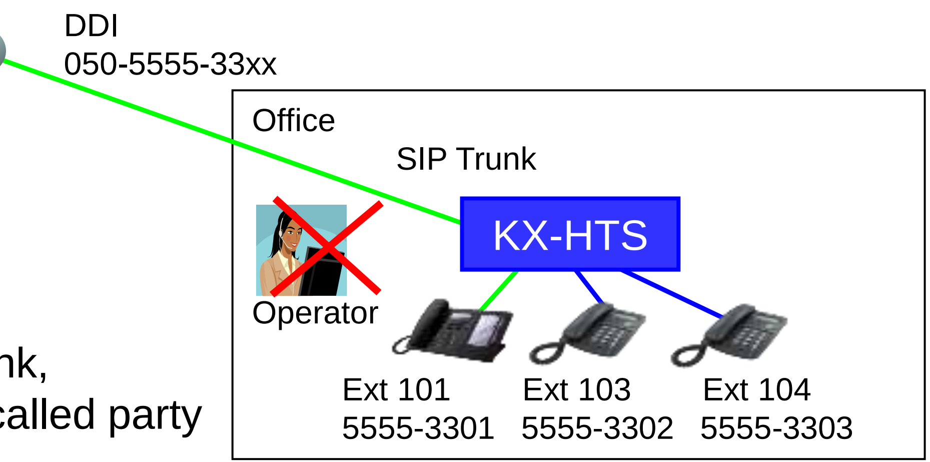
PBX SE team

1. Overview

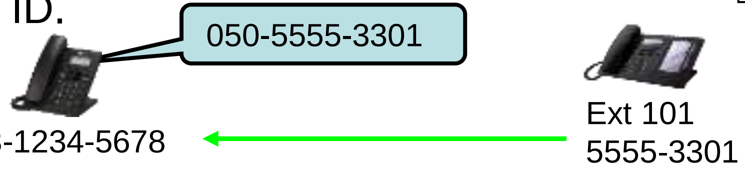
SIP trunk is useful in order to save cost.
KX-HTS supports 200 DDI numbers for one or two SIP carriers.
Extension can be called directly by DDI without operator.





CLIP

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



2. Table of Contents

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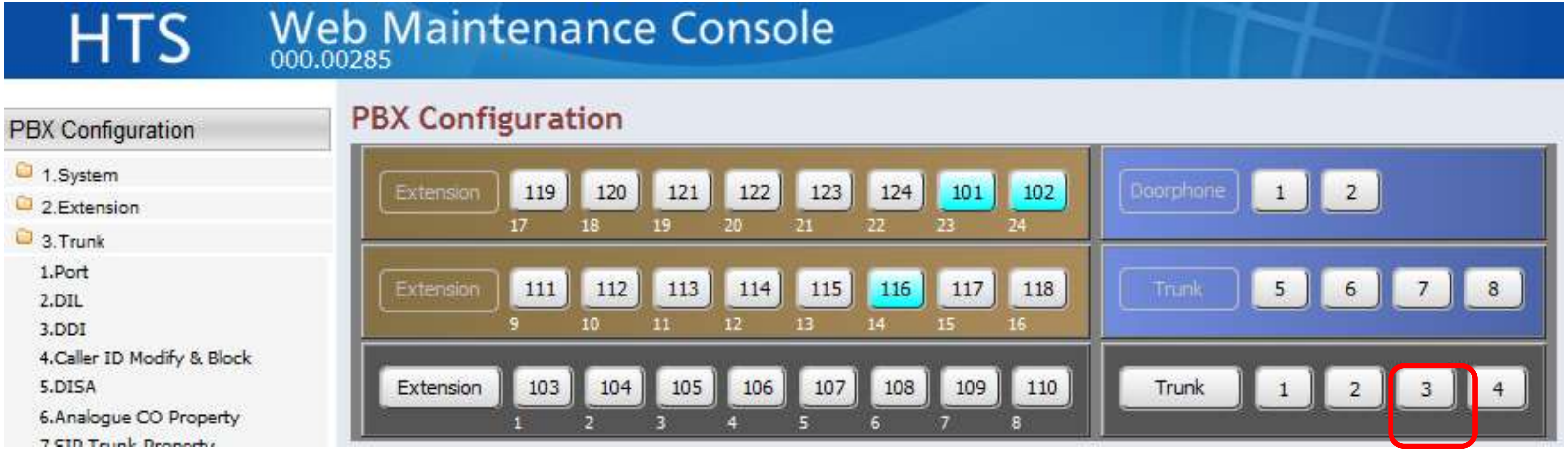
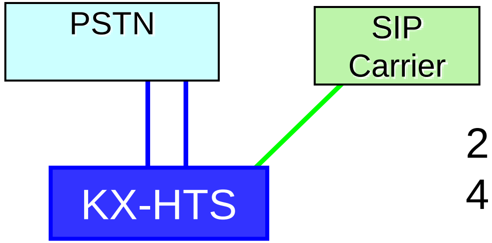
1	SIP trunk programming
2	Test after SIP trunk programming
3	ARS Programming
4	Peripheral Settings (NAT, Routing)

Chapter 1
SIP Trunk Programming

11. Click Port to change to SIP.

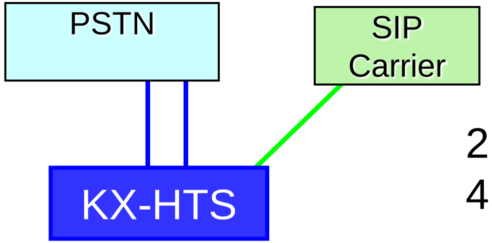
Click port 3 for example.





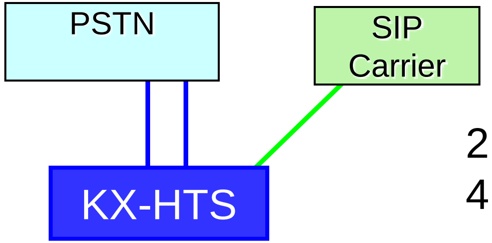
2 lines of Analog CO and
4 ch of SIP for example.

12. Change the port to SIP.



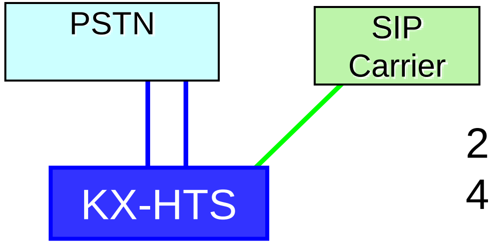
2 lines of Analog CO and
4 ch of SIP for example.

13. Assign Trunk Group.



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

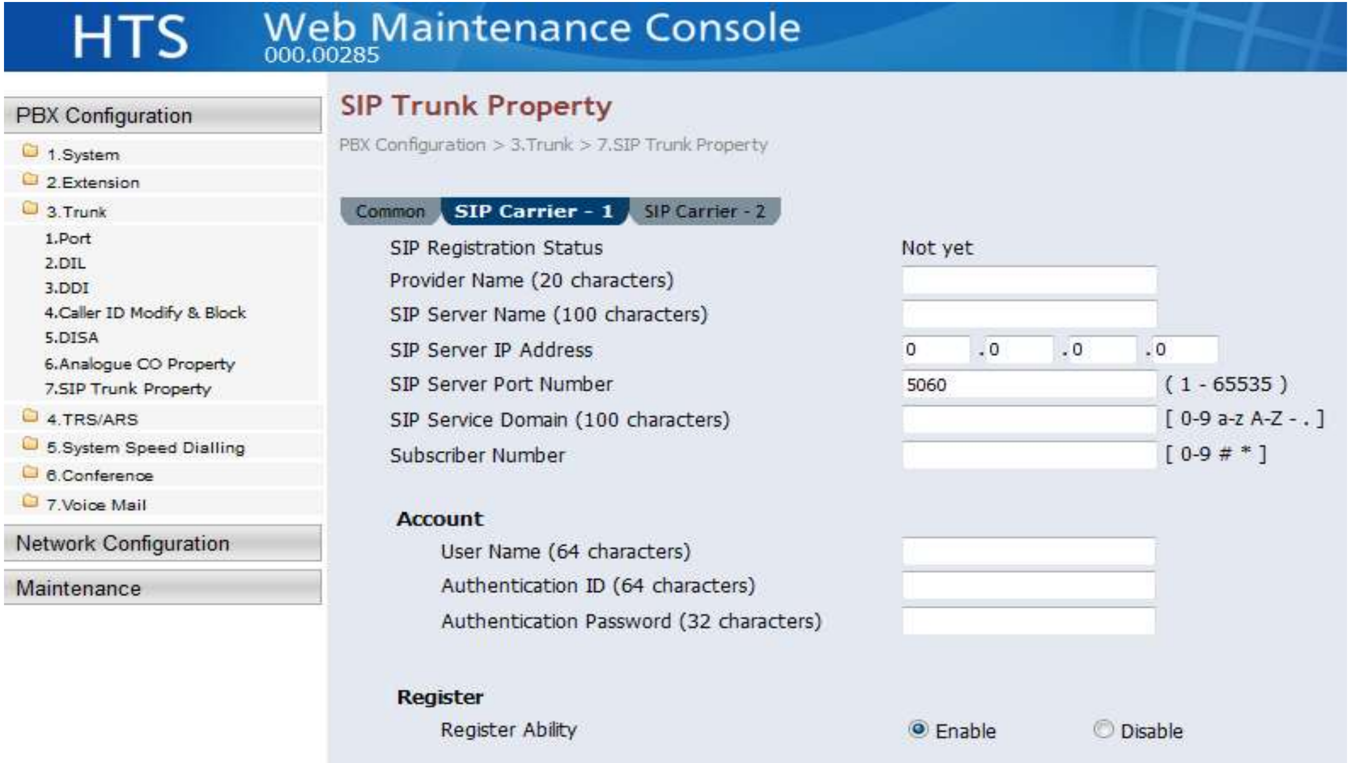
14. Repeat to other trunk.



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

15. Assign SIP account.

Ask your account to your SIP carrier.

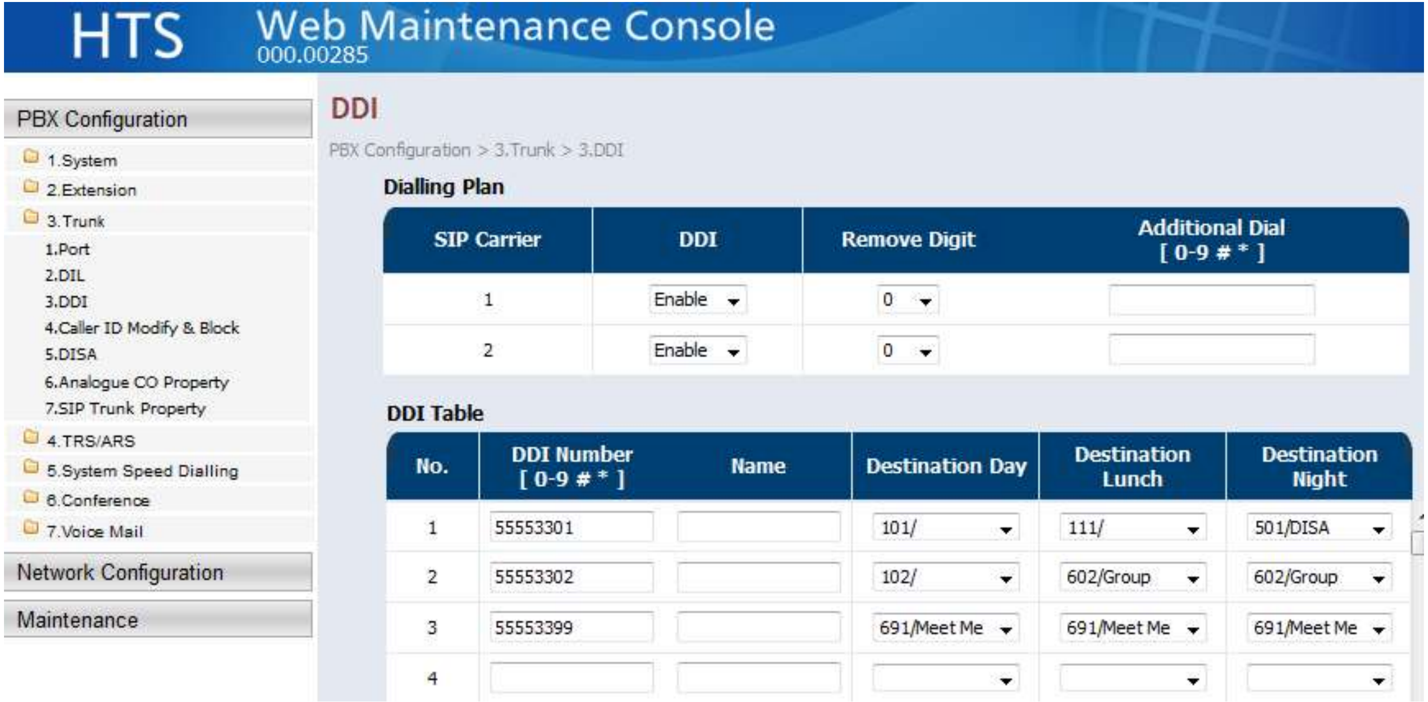


16. Assign Voice Codec.

Ask correct CODEC to your SIP carrier.

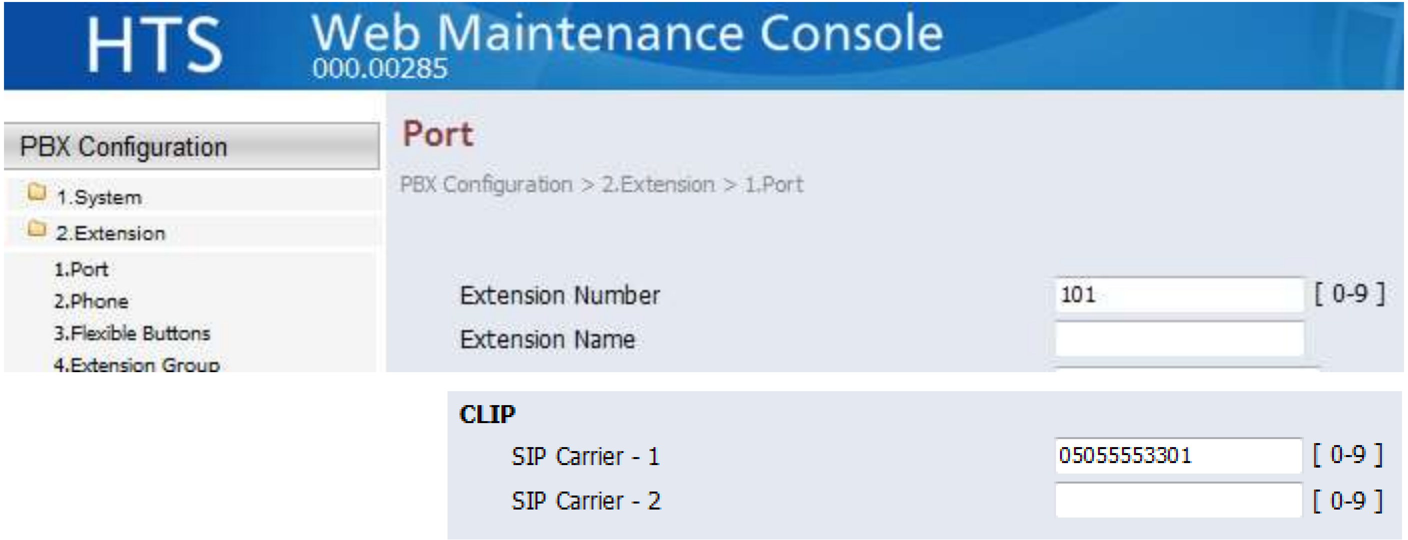
If it is G.729, assign G729 for 1st priority and None for 2nd and 3rd priority.

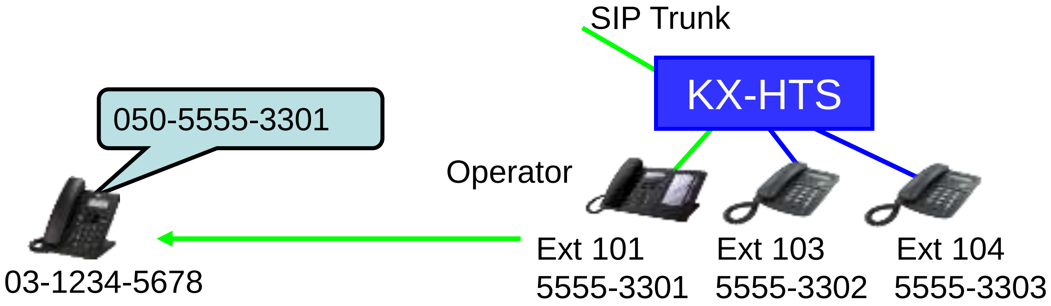
18. Assign DDI.



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.

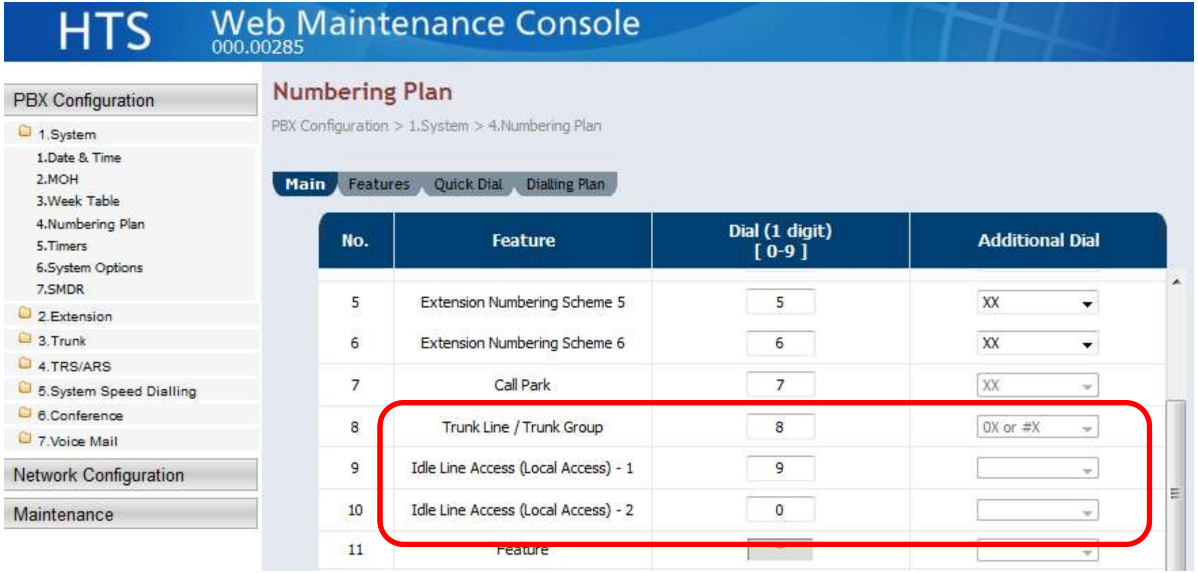
19. Assign CLIP.

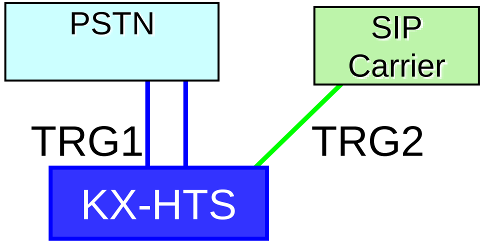




Chapter 2
Test after SIP Programming

21. Confirm CO Access Number.



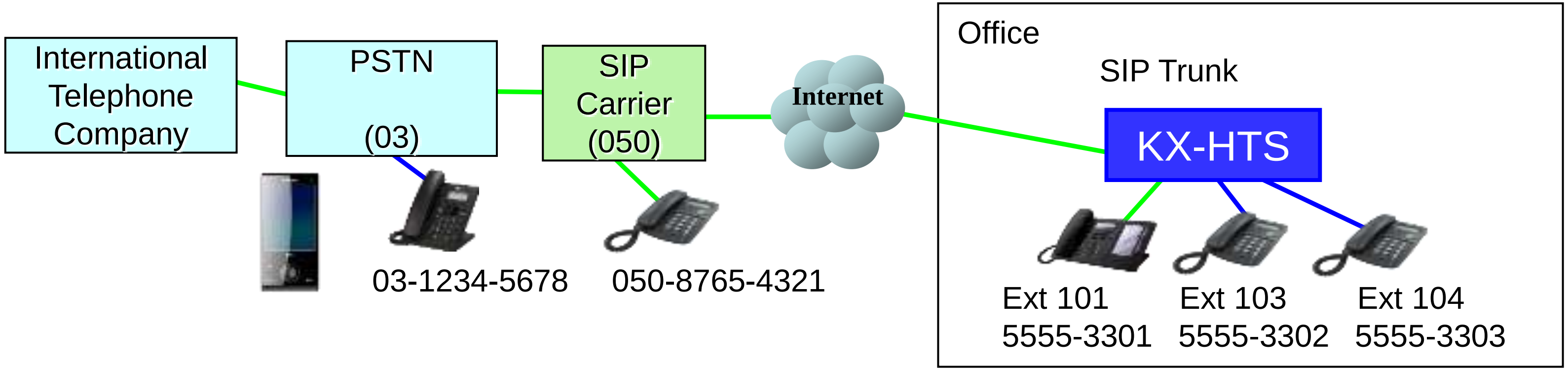


Dial 0 or 9 for any trunk + Phone number.
Dial 8#1 for TRG1 + Phone number.
Dial 8#2 for TRG2 + Phone number.

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

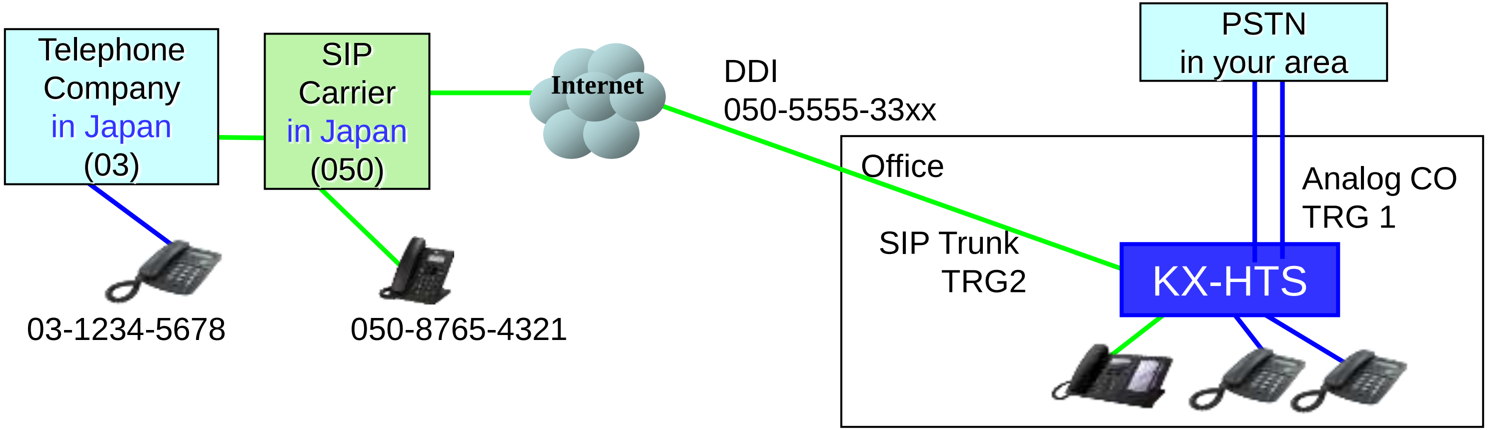
SIP trunk can be selected automatically for programmed phone number. Dialed number can be modified automatically.

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

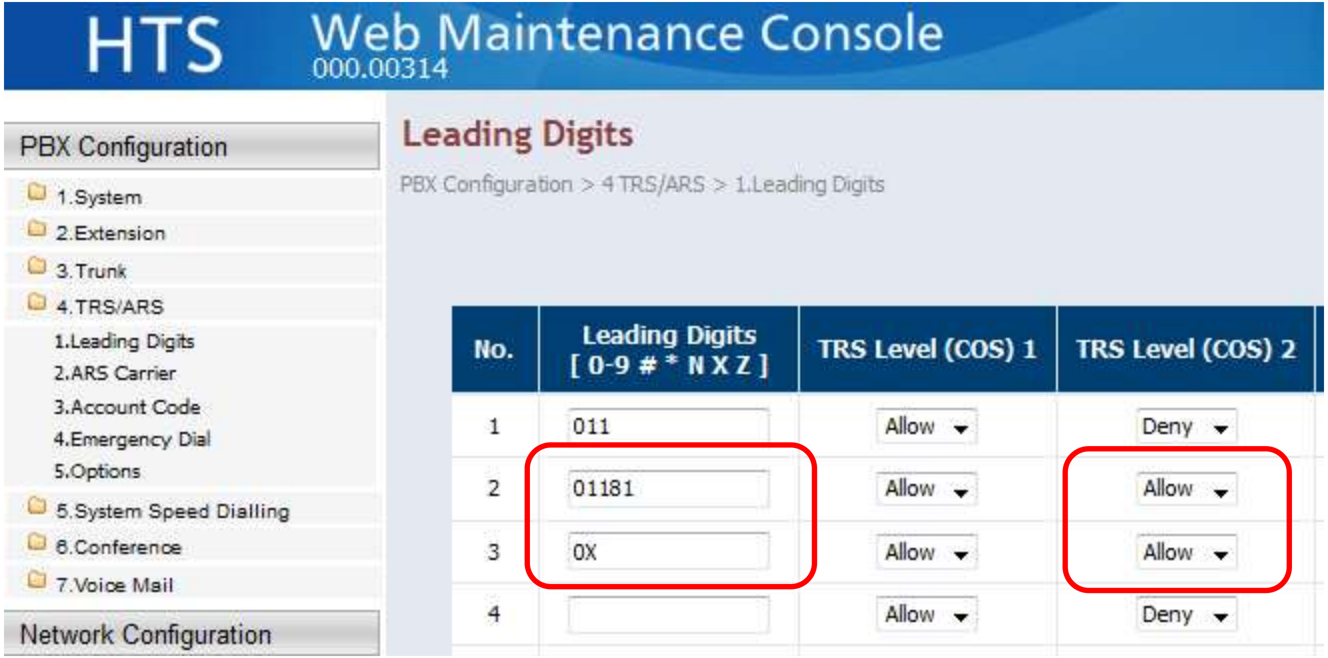
Dialed number by user : 9/0-011-81-3-1234-5678

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

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



31. Assign Leading Digits & Change TRS.



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.

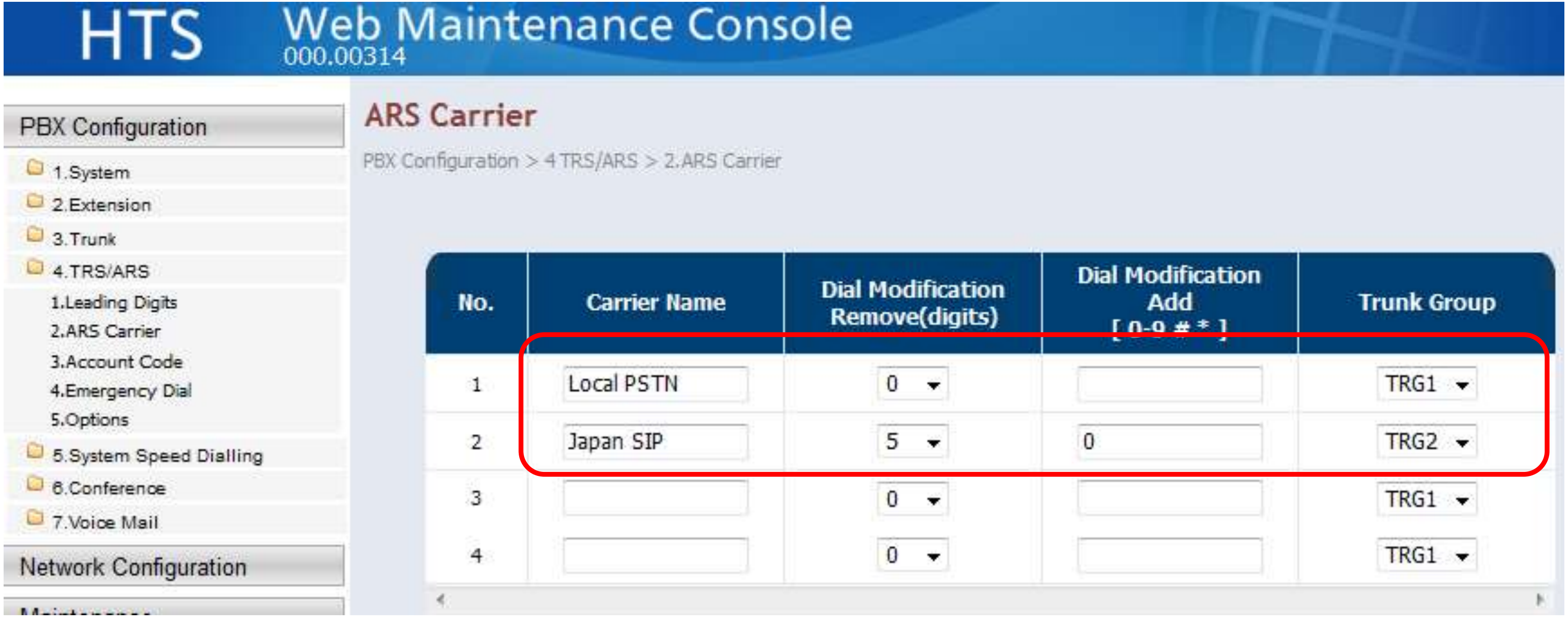
No.	Leading Digits [0-9 # * N X Z]	TRS Level (COS) 1	TRS Level (COS) 2	ARS Carrier Priority-1	ARS Carrier Priority-2	ARS Carrier Priority-3
1	011	Allow	Deny	1:	None	None
2	01181	Allow	Allow	2:	1:	None
3	0X	Allow	Allow	1:	None	None
4		Allow	Deny	None	None	None

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.

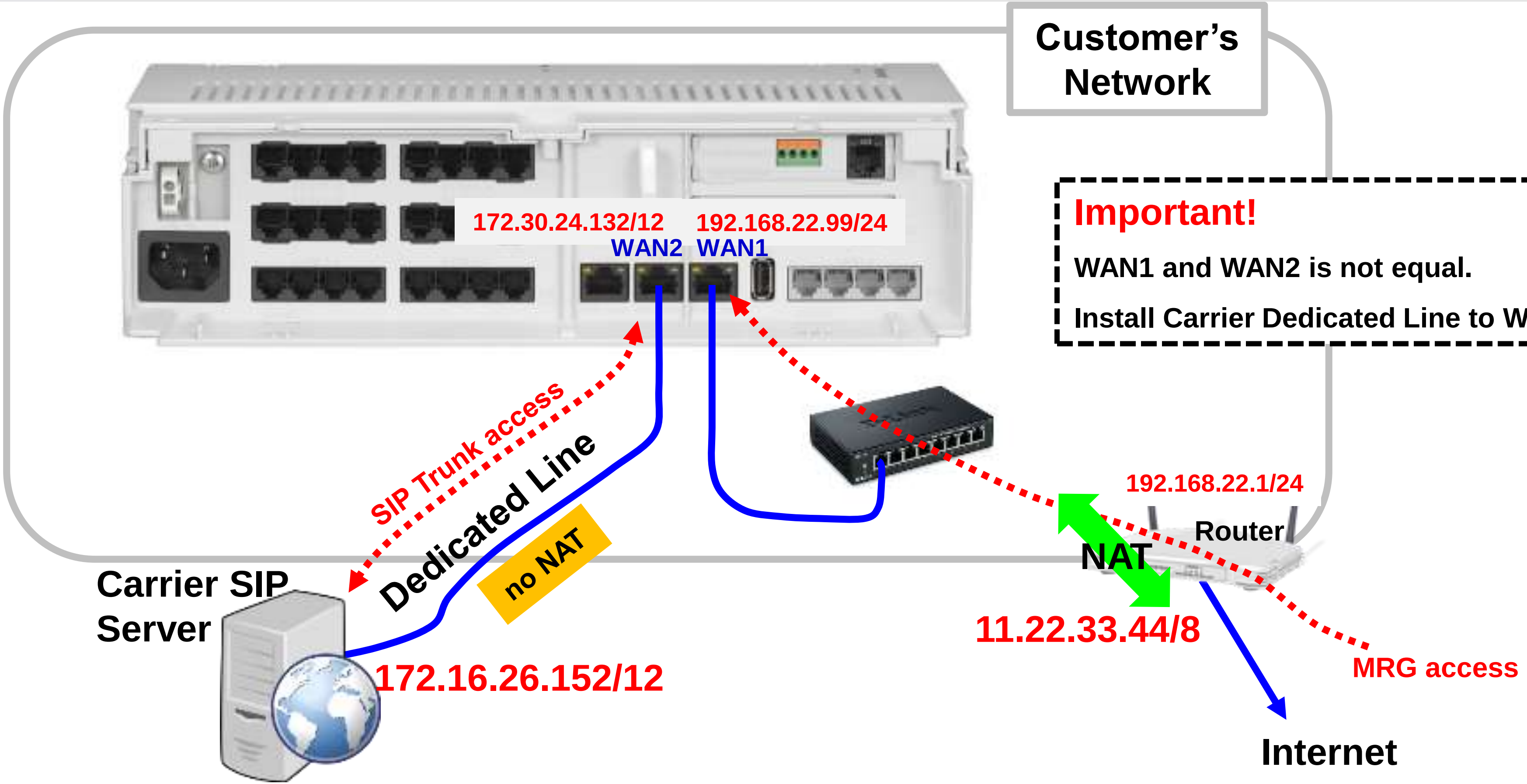
Remove 5 digits (01181) and add “0”.



Programming is completed. Try making call by ARS.

Chapter 4
Peripheral settings (NAT, Routing)

41. Diagram



Important!

WAN1 and WAN2 is not equal.

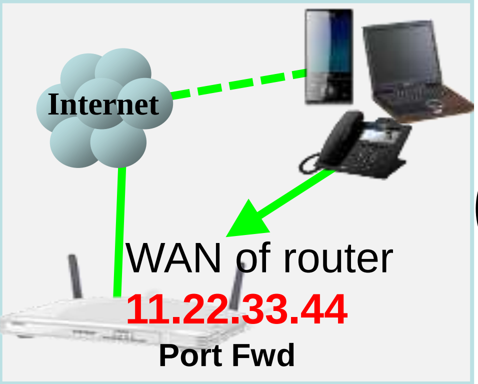
Install Carrier Dedicated Line to WAN2.

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 (ms)
NAT Traversal	Off / Fixed IP Address / STUN / 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!** OK!

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	
. . .	. . .	. . .		Add
172.16.0.0	255.255.240.0	172.30.0.1	WAN2_DHCP_Client	Delete

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
○	WAN1_Fix_IP	WAN1	Fix IP	◉
○	WAN2_DHCP_Client	WAN2	DHCP Client	○

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.