

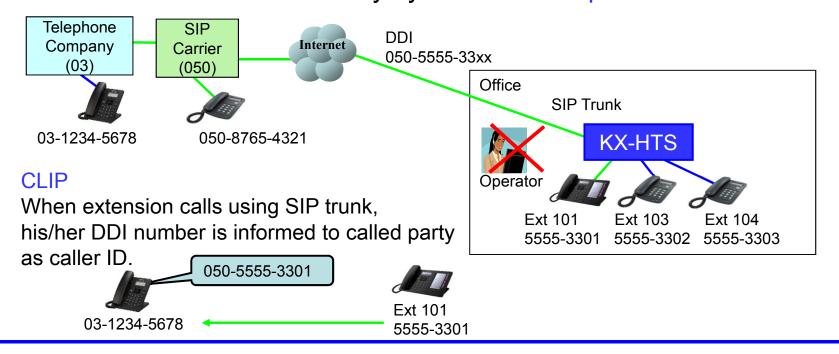
Panasonic System Networks PBX SE team

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.



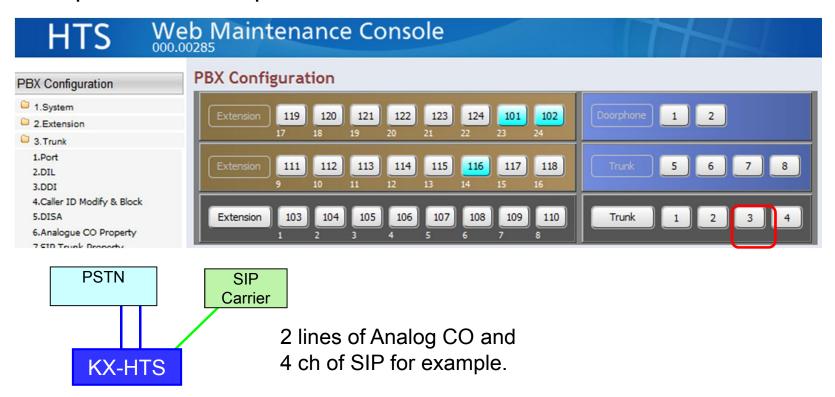
2. Table of Contents

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Chapter 1 SIP Trunk Programming

11. Click Port to change to SIP.

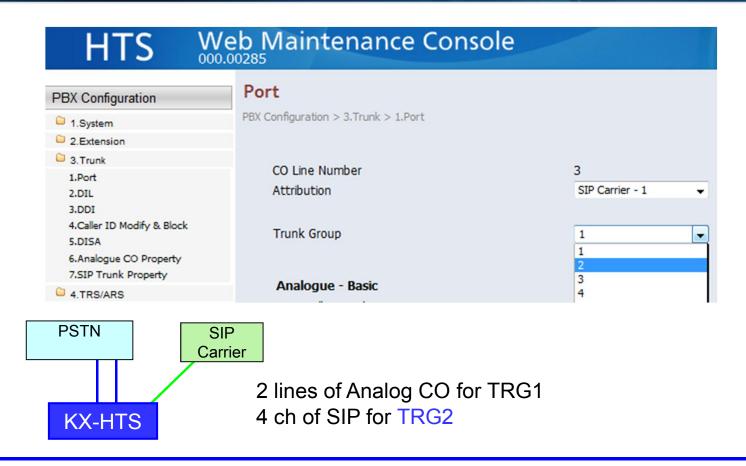
Click port 3 for example.



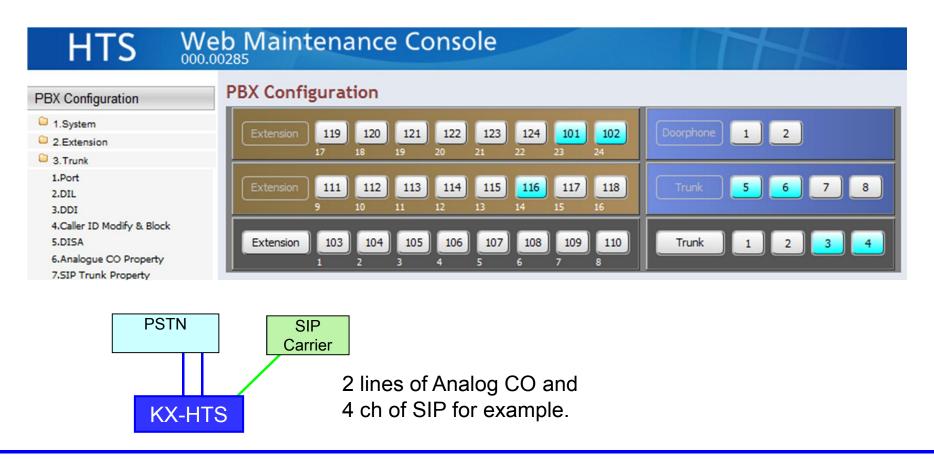
12. Change the port to SIP.



13. Assign Trunk Group.

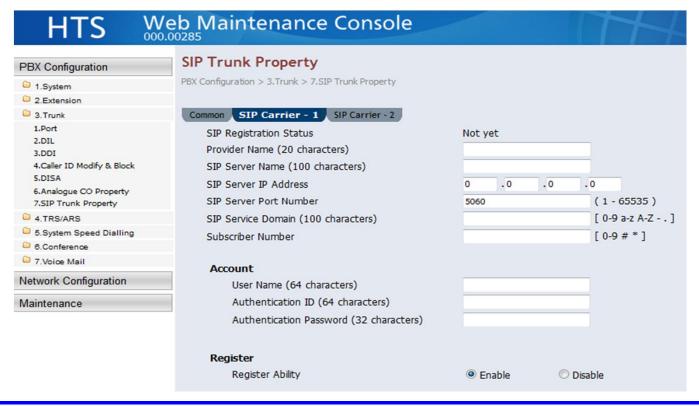


14. Repeat to other trunk.



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.

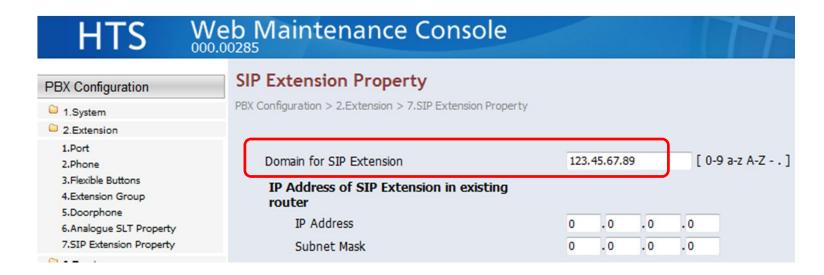
PBX Configuration > 3.Trunk > 7.SIP Trunk Property Voice G.711A IP Codec Priority 1st © G.711Mu O G.729A © G.711A **⊙** G.711Mu IP Codec Priority 2nd O G.729A O None © G.711A ○ G.711Mu IP Codec Priority 3rd O G.729A None

17. Assign SIP Domain Name.

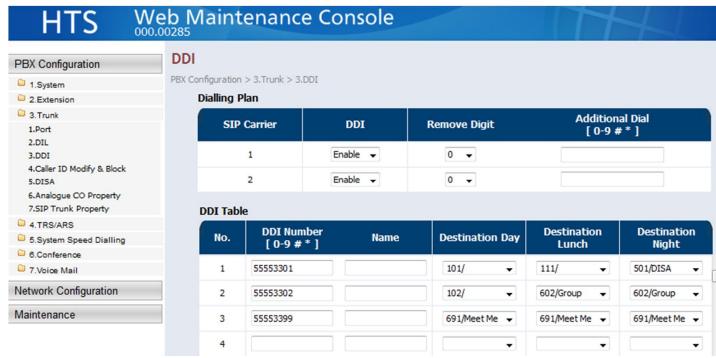
SIP domain name is required for call using WAN.

123.45.67.89 is example.

Some SIP phone does not support character (a-z A-Z) for SIP domain name.

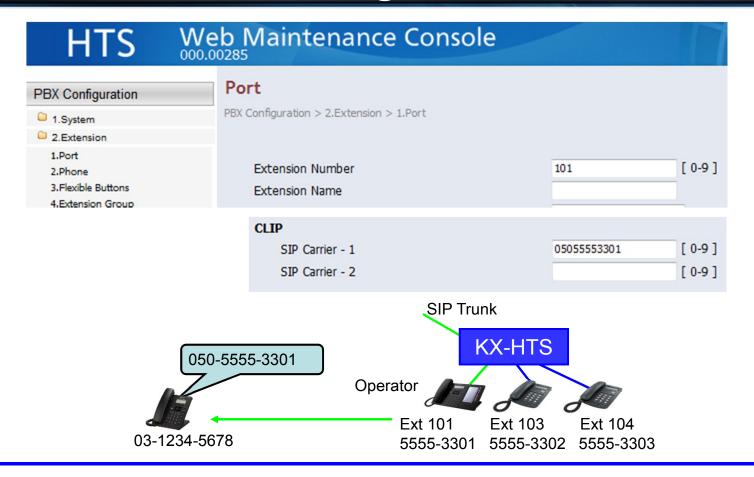


18. Assign DDI.



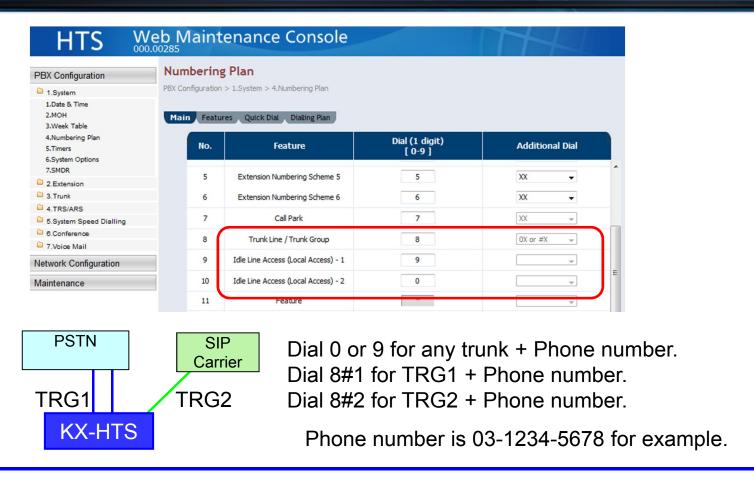
When received DDI number after modification by "Remove" and "Add" cannot be found in DDI table, but it is same as extension number, destination becomes the extension. If it is not extension number also, DIL is applied.

19. Assign CLIP.



Chapter 2 Test after SIP Programming

21. Confirm CO Access Number.

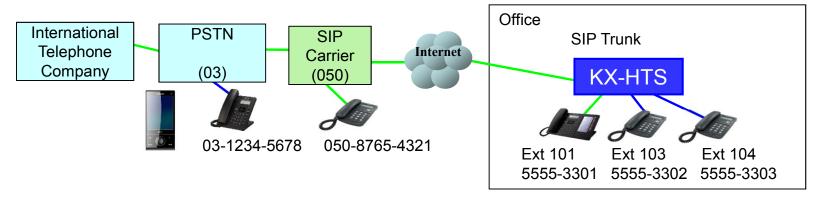


22. Make and Receive a call actually.

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	
Call to	needs)	

Chapter 3 ARS Programming

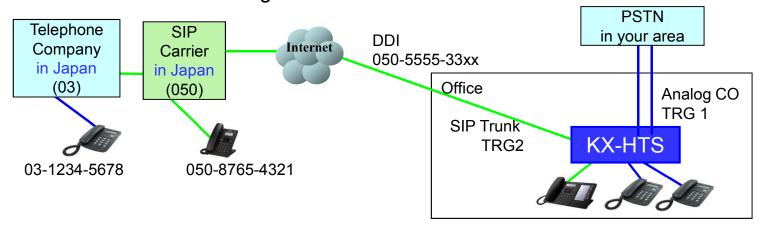
30. ARS: Automatic Route Selection

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.

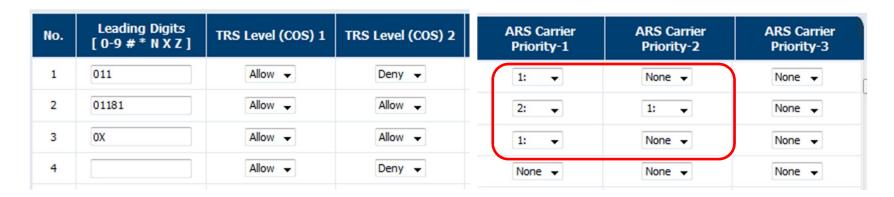


31. Assign Leading Digits & Change TRS.



Default of TRS level is 2 for all extensions.

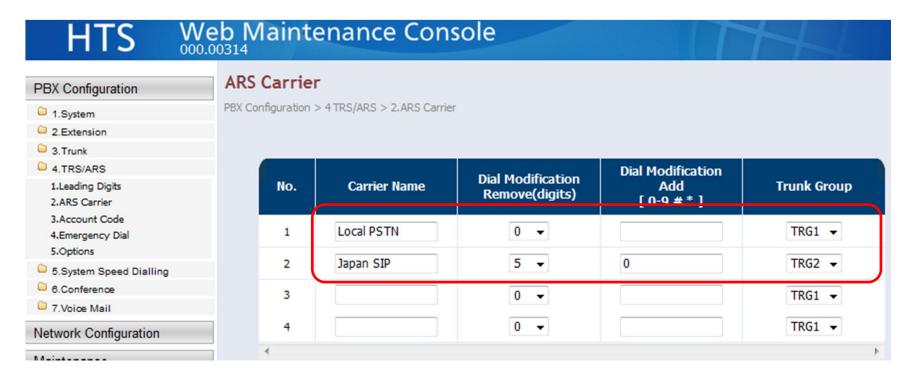
32. Assign ARS Carrier.



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.



Programming is completed. Try making call by ARS.

Thank you!

https://namlong.vn Hotline: 092 888 2345

Revision

Date	No.	Change
July 28, 2016	All	First official release
August 8, 2016	Chapter 3	ARS was added.