Cisco ISDN PRI to SIP Gateway

Supported features

- Full ISDN E1 emulation
- Early media support
- Inbound calling. Type SIP REGISTERED TRUNK
- Outbound Calling
- ISDN PRI equivalent
- Secure Calling via SIP Encrypt platform available

Cisco Platforms ratified:

- Cisco 2800, 3800
- Cisco 2900, 3900
- Cisco AS5300
- Cisco AS5350XM
- Cisco AS5400
- Cisco AS5400XM
- Cisco UC560 T1/E1 varient (Single E1 only) **

Cisco router chassis requires PVDM resource, voice enabled E1 cards and voice feature enabled software / licence.

UC560-T1/E1 is a ‘Small Business’ PBX and contains a PVDM2-64 and a single E1 card. This device has all the components needed for an ISDN to SIP Gateway, (or as a voice gateway for the BE6000).

E1 presentation of Cisco E1 line cards is wired as “CPE” side. To emulate the ISDN Network at Layer1 – use an E1 cross over cable made by swapping pins 1,4 and 2,5
Example configuration snippits were configured on a 2821 router with x2 VWIC2-2MFT-T1/E1 and x3 PVDM2-64. Recommended IOS version is 12.4(24)T3 and later.

Step one: Configure VoIP.co.uk portal for inbound calling type: SIP REGISTERED TRUNK

Create a SIP account for the Cisco router. A SIP Account is a username / password pair which a SIP phone / endpoint uses to authenticate itself.

1. On the my.voip.co.uk portal, click on SIP-AOR and create a SIP account for the Cisco router. Give the SIP account a meaningful name – like "My Cisco gateway". Also make sure the password is complex.
2. Click on dashboard and then "Incoming targets". Create a New Incoming Target; TYPE: SIP_REGISTERED TRUNK and again call the incoming target something meaningful – "such as Route calls to my Cisco gateway". It is recommended that the alphanumeric characters "AAA" are pre-pended to an incoming call to facilitate call routing on the ISDN/SIP gateway. The Registered Trunk should contain the user template: "AAA${e164}"
3. Click on the new incoming target and add in the new SIP account you just created.
4. Click on Dashboard / Phone numbers and configure a telephone number to route calls to your new incoming target.

Step Two. Setup E1 cards

1. Most Cisco E1 cards are T1/E1 capable and they need to be set for E1 (Europe)

```
!
card type e1 0 0
card type e1 0 1
!
network-clock-participate wic 0
network-clock-participate wic 1
network-clock-select 5 E1 0/0/0
network-clock-select 6 E1 0/0/1
!
```

Step Three. Create 30 channels on each E1

```
!
controller E1 0/0/0
pri-group timeslots 1-31
!
controller E1 0/0/1
pri-group timeslots 1-31
!
controller E1 0/1/0
pri-group timeslots 1-31
!
controller E1 0/1/1
pri-group timeslots 1-31
!
interface Serial0/0/0:15
description “use dial-peer voice 10″
no ip address
encapsulation hdlc
isdn switch-type primary-net5
isdn timer T309 400000
isdn protocol-emulate network
isdn incoming-voice modem
isdn send-alerting
isdn sending-complete
no cdp enable
!
interface Serial0/0/1:15
description “use dial-peer voice 11″
no ip address
encapsulation hdlc
isdn switch-type primary-net5
isdn timer T309 400000
isdn protocol-emulate network
isdn incoming-voice modem
isdn send-alerting
isdn sending-complete
no cdp enable
!
interface Serial0/1/0:15
description “use dial-peer voice 12″
no ip address
encapsulation hdlc
```

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isdn switch-type primary-net5
isdn timer T309 400000
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interface Serial0/1/1:15
description “use dial-peer voice 13″
no ip address
encapsulation hdlc
isdn switch-type primary-net5
isdn timer T309 400000
isdn protocol-emulate network
isdn incoming-voice modem
isdn send-alerting
isdn sending-complete
no cdp enable
!
```

Step 4 – Configure region cadence

```
!
voice-port 0/0/0:15
cptone GB
bearer-cap Speech
!
voice-port 0/1/0:15
cptone GB
bearer-cap Speech
!
voice-port 0/0/1:15
cptone GB
bearer-cap Speech
!
voice-port 0/1/1:15
cptone GB
bearer-cap Speech
!
```

Step 5 – Configure router with Basic SIP settings

```
!
sip-ua
authentication username <SIP username created in Step 1> password <SIP password in step 1>
credentials username <SIP username created in Step 1> password <SIP password in step 1>
realm proxy.voip.co.uk
retry invite 2
registrar dns:proxy.voip.co.uk expires 120
sip-server dns:proxy.voip.co.uk
!
voice class codec 1
codec preference 1 g711alaw
codec preference 2 g711ulaw
codec preference 3 g729r8
!
```

Step 6 – ISDN to SIP call routing

Inbound POTS dial-peer to accept calls from ISDN

```
!
dial-peer voice 9 pots
description incoming pots dial-peer
translation-profile incoming isdn-to-voip
incoming called-number .+
direct-inward-dial
no sip-register
!
```

Outbound VoIP dial-peer to route calls to VoIP.co.uk

```
!
dial-peer voice 1 voip
description “Outbound SIP”
destination-pattern ^…T
voice-class codec 1
session protocol sipv2
session target sip-server
dtmf-relay rtp-nte
fax-relay ecm disable
fax nsf 000000
fax protocol t38 ls-redundancy 3 hs-redundancy 1 fallback pass-through g711alaw
```

no vad
!
Step 7 – SIP to ISDN
Inbound VoIP dial-peer to accept calls from VoIP.co.uk

dial-peer voice 997 voip
description incoming voip
rtp payload-type cisco-codec-fax-ind 124
voice-class codec 1
session protocol sipv2
session target sip-server
incoming called-number ^AAA.+
dtmf-relay rtp-nte
fax-relay ecm disable
fax nsf 000000
fax protocol t38 ls-redundancy 3 hs-redundancy 1 fallback pass-through g711alaw
no vad
!

Outbound POTS dial-peer to route calls to the ISDN PBX:

!
dial-peer voice 10 pots
description POTS talking dial peer for E1 #0
translation-profile outgoing voip-to-isdn
preference 1
destination-pattern ^AAA.+
port 0/0/0:15
no sip-register
!
dial-peer voice 11 pots
description POTS talking dial peer for E1 #1
translation-profile outgoing voip-to-isdn
preference 1
destination-pattern ^AAA.+
port 0/0/1:15
no sip-register
!
dial-peer voice 12 pots
description POTS talking dial peer for E1 #2
translation-profile outgoing voip-to-isdn
preference 1

```
destination-pattern ^AAA.+
port 0/1/0:15
no sip-register
!
dial-peer voice 13 pots
description POTS talking dial peer for E1 #3
translation-profile outgoing voip-to-isdn
preference 1
destination-pattern ^AAA.+
port 0/1/1:15
no sip-register
!
```

IOS 15.0 and later (Toll fraud feature)

```
voice service voip
ip address trusted list
ipv4 193.203.210.0 255.255.254.0
```

Step 8 – Set up number translations

```
voice translation-rule 501
rule 1 /^44\(.+\)$/ /\1/ type any national plan any isdn
rule 3 /^\+44\(.+\)$/ /\1/ type any national plan any isdn
rule 10 /\(.+\)/ /0\1/ type any national plan any isdn
!
voice translation-rule 601
rule 1 /^AAA44\(.+\)$/ /\1/ type any national plan any isdn
voice translation-profile voip-to-isdn
translate calling 501
translate called 601
!
```

Note: translation-rule 601 presents the 10 digit phone number to the PBX. It is normal in the UK for BT to present either the full number minus the leading zero (cities) or the trailing 6 or 4 digits.

Example: Phone number AAA1869222500 arrives on the gateway

```
TDM_Gateway#test voice translation-rule 601 AAA441869222500
Matched with rule 1
Original number: AAA441869222500        Translated number: 1869222500
```

Original number type: none   Translated number type: national
Original number plan: none   Translated number plan: isdn
TDM_Gateway#

To present the trailing six digits, rule 601 would be re-written:

voice translation-rule 601
rule 1 /^AAA441869\(.+\)$/ /\1/ type any national plan any isdn

Step 9 – Test

Test inbound and outbound telephony. Check:

DTMF operation

Early Media Rinback tones “ring ring”

Some older phone systems required an “ISDN alerting” message to be sent by the gateway. Unless specifically enabled the Ringback tone may be not heard by the PBX user when dialling out.

!

voice call send-alert

!