PCS Partner & Centralised Partner Interface

Introduction

The PCS Partner Interface is a special TCP connection to a Splicecom system client that allows for external control of calls and additional special call handling. In effect it allows access to all of the features a user can directly access as a client. In addition a number of other special features may be controlled such as conferencing.

The PCS Partner Interface is accessed on TCP port 50xx where xx is the port number. 4330 and 5330 Phone Modules are numbered 1 to 30 (or 1 to 15 in the case of 4315/5315 and 1 to 8 in the case of the P308 Intelligent Gateway Module) and are thus port 5001 to 5030, 5015 & 5008 respectively. PCS 5xx, PCS 410/400 & PCS 100 IP Phones and Navigate/PCS 60/PCS 50 IP Softphones only have a single port and are 5001 only.

All input is newline terminated (cr or lf) and does not rely on packet boundaries. There are a number of CSV (Comma Separated) Commands that can be executed and these are specified in "Commands".

A number of output messages can be generated and these are documented in "Outputs"

In general the <partnerid> and <localid> should be treated as strings and reproduced exactly as output. The partner id is an id generated by you the partner and is used to allow a unique identification for your application. The <localid> is the clients view of the world and is a temporal unique id. (Please be aware these ids can be re-used over time and thus a partner should rely on the pair if possible).

Centralised Partner allows Splicecom's SelectVoice platforms to provide similar functionality with Yealink phones and is essentially the PCS Partner interface with the following changes:

- You must use a TLS connection to the SelectVoice server on port 4018.
- You must login with a user and password, instead of a partner code, and the LOGIN command must have a comma after the password, e.g. LOGIN,2001,1234,

Splicecom reserves the right to make alterations or amendments to the detailed specifications at its discretion.

INTRODUCTION 1

OUTPUT 4

UPDATECOMPLETE 4
NEWCALL 4
CALLOK 4
INSETUP 4
INSIGNAL 5
INSETUPACK 5
INPROCEEDING 5
INPROGRESS 5
INALERTING 6
INCONNECT 6
INRELEASE 6
OUTALERTING 7
OUTCONNECT 7
OUTRELEASE 8
OUTDIGITS 8
OUTSETUP 8
CONFUPDATE 8
CURRENTCALL 9
REMOVECALL 9
LOGGED IN AS 9
NOT LOGGED IN 9
CALLINFO 10
AUTO RECORD 10
OUT OF OFFICE 10
ACCOUNT CODING 10
AUTO ANSWER 11
RING TUNE 11
FORCED WEB PAGE 11
PHONE RECORD CHANGED 11
USER RECORD CHANGED 12
UPDATE ALL INFO 12
NOTE AND HISTORY AVAILABLE 12
NOTE ADDED TO CALL 12
AUTO CONFERENCE 12
UPDATE USER MESSAGES 13
HOOK STATE 13
AUTO LISTEN 13
BLANK 13
UNBLANK 14

COMMANDS 15

LOGIN 15
LOGOUT 15
NEWCALL 15
DELCALL 15
DIAL 15
DIALDIGITS 16

CONNECT 16
CONNECT_TONE 16
PARK 16
PICKUP 17
TRANSFER 17
WRAPUPDONE 17
SENDTOVOICEMAIL 17
CURRENTCALL 18
ACCOUNTCODE 18
RECORD 18
CONFERENCE 18
BLANK 19
UNBLANK 19

APPENDIX 1 .... **20**

Output

UPDATECOMPLETE
UPDATECOMPLETE

Sent to indicate that all the key information has been sent after the initial connection.

NEWCALL
NEWCALL,<partnerid>,<callid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

A call has been created ready for use.

CALLOK
CALLOK,<partnerid>,<callid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

A call has been given permission to send the setup or be answered. You may now DIAL. (H.323 ARQ)

inSetup
inSETUP,<partnerid>,<callid>,<cn>,<telephone num>,<local cn>,<local num>,<ring tune>,<ooo>,<ooo_msg>,<confid>,<autoanswer>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<cn> = name of source
<telephone num> = telephone number of source
<local cn> = name of local target
<local num> = number of local target
<ring tune> = number of the ring tune to play
<ooo> = out of office mode
<ooo_msg> = out of office text
<confid> = system call identifier
<autoanswer> = Whether the call is to be automatically answered

An incoming call has been received.

inSignal
inSIGNAL,<partnerid>,<callid>,"<digits>"

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<digits> = digits/alpha from other end

A inband signal has been received on a call. Non-displayable digits will be sent in octal.

inSetupAck
inSETUPACK,<partnerid>,<callid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

Overlapped dialling has started. You can now send DIALDIGITS.

inProceeding
inPROCEEDING,<partnerid>,<callid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

An outgoing call is now proceeding thru the network and all dialling has been completed.

inProgress
inPROGRESS,<partnerid>,<localid>,<reason>,<progressInfo>,<progressLocation>

<reason> = See H.323 Reason Codes defined under inRelease
<progressInfo> = 1 – Call not end to end ISDN (may be further inband information. Call cutthru)
2 – Destination not ISDN
3 – Origination not ISDN
4 – Call returned to ISDN
8 – Inband information available (Call cutthru)

Only type 1 and 8 will be cutthru. The others are for information only.

<progressLocation> = 0 – User
1 – Private network serving local user
2 – Public network serving local user
3 – Transit network
4 – Public network serving remote user
5 – Private network serving remote user
7 – International network
10 – Network beyond interworking point

Establishment of an outgoing call is in progress.

inAlerting

inALERTING,<partnerid>,<callid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

An outgoing call is now ringing at the remote end.

inConnect

inCONNECT,<partnerid>,<callid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

An outgoing call is now connected. The remote end has answered.

inRelease

inRELEASE,<partnerid>,<callid>,<reason>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<reason> =

```
#define H323_REASON_OFFSET          0x1000

        enum ReleaseReason
        {
/*              // Q.931 Cause Code
                noRouteToDestination = 3,
                normalCallClearing = 16,
                userBusy = 17,
                subscriberAbsent = 20,
                invalidNumberFormat = 28,
                normalUnspecified = 31,
                noCircuitAvailable = 32,
                networkOutOfOrder = 38,
                temporaryFailure = 41,
                switchingEquipmentCongestion = 42,
                resourceUnavailable = 47,
                incompatibleDestination = 88,
                interworkingUnspecified = 127,
*/
     noReason = 0,
```

```
        // H.323 ReleaseCompleteReason
        noBandwidth = H323_REASON_OFFSET,
        gatekeeperResources,
        unreachableDestination,
        destinationRejection,
        invalidRevision,
        noPermission,
        unreachableGatekeeper,
        gatewayResources,
        badFormatAddress,
        adaptiveBusy,
        inConf,
        undefinedReason,
        //...
        facilityCallDeflection,
        securityDenied,
        calledPartyNotRegistered,
        callerNotRegistered,
        newConnectionNeeded,
        nonStandardReason,
        replaceWithConferenceInvite,
        genericDataReason,
        neededFeatureNotSupported,
        tunnelledSignallingRejected,
        invalidCID,
    };
```

A call has now cleared down. The remote end cleared the call.

Note: A full list of cause codes and descriptions can be found in Appendix I.

outAlerting
outALERTING,<partnerid>,<callid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

An incoming call is now ringing at the terminal.

outConnect
outCONNECT,<partnerid>,<callid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

An incoming call has been answered by the terminal.

outRelease

outRELEASE,<partnerid>,<callid>,<reason>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<reason> = Reason for the call to be disconnected.
If below 1000, the number is a Q.931 cause code, otherwise it translates to an H.323 release reason.

A call has been cleared by the local terminal.

outDigits

outDIGITS,<partnerid>,<callid>,<digits>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<digits> = digits being dialled

The terminal is dialling digits.

outSetup

outSETUP,<partnerid>,<callid>,<cn>,<telephone num>,<local cn>,<local telephone num>,<confid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<cn> = name of party being called
<telephone num> = number of party being called
<local cn> = name of source
<local telephone num> = number of source
<confid> = system call identifier

The terminal is dialling a new call.

CONFUPDATE

CONFUPDATE,<partnerid>,<callid>,<recording>,<conf partnerid>,<conf localid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<recording> = is currently recording
<conf partnerid> = call that is conferenced in or 0
<conf localid>

Conference update.

CURRENTCALL

CURRENTCALL,<partnerid>,<callid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

This is now the currently active call (the one the terminal can hear).

REMOVECALL

REMOVECALL,<0>,<callid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

This call has been removed and no longer exists.

LOGGED IN AS

LOGGED IN AS,<user guid>,<ldap_usernam>,<ldap_password>,<ldap_server>,<phone dn>,<gatekeeper id>,<product_version>,>device_type>,<company_guid>

<user guid> = GUID of user logged in as.
<ldap username> = name of ldap user
<ldap password> = password for ldap access
<ldap server> = server to use for ldap requests
<phone dn> = id of phone partner'd to
<gatekeeper id> = id of the gatekeeper system/module
<product version> = version of the product attached to
<device type> = type of the device attached to
<company_guid> = GUID of the company that the user belongs to, if applicable

The terminal is logged in as… This is then followed by CALLINFO

NOT LOGGED IN

NOT LOGGED IN

CALLINFO

CALLINFO <partnerid>,<localid>,<cn>,<telephone>,<local cn>,<local telephone>,<ring tune>,<ooo>,<ooo_msg>,<state>,<transferring>,<incoming>,<recording>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<cn> = name of other end
<telephone> = number of other end
<local cn> = name of this end
<local num> = number of this end
<ring tune> = ring tune number
<ooo> = out of office mode
<ooo_msg> = out of office text
<state> = state of call
<transferring> = is call in the process of transferring (hide)
<incoming> = in or outgoing call
<recording> = call is currently recording

AUTO RECORD

AUTO RECORD,<partnerid>,<callid>,<autorecordnumber>,<mode>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<autorecordnumber> = number to record to (id in voicemail)
<mode> = autorecord mode

An autorecord command has been received.

OUT OF OFFICE

OUT OF OFFICE,<partnerid>,<callid>,<ooo>,<ooo_msg>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<ooo> = out of office mode
<ooo_msg> = out of office text

The out of office message is.

ACCOUNT CODING

ACCOUNT CODING,<partnerid>,<callid>,<mode>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<mode> = account coding mode (blank)

Account coding is required. You should prompt the user to enter the account code.

AUTO ANSWER

AUTO ANSWER,<partnerid>,<callid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

Auto answer mode has been selected.

RING TUNE

RING TUNE,<partnerid>,<callid>,<tune>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<tune> = ring tune number

Set the ring tune to the number defined

FORCED WEB PAGE

FORCED WEB PAGE,<partnerid>,<callid>,<url>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<url> = url of web page

A web page needs to be displayed

PHONE RECORD CHANGED

PHONE RECORD CHANGED,<partnerid>,<callid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

The configuration for the phone has been updated.

USER RECORD CHANGED
USER RECORD CHANGED,<partnerid>,<callid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

The coniguration for the user has been updated.

UPDATE ALL INFO
UPDATE ALL INFO,<partnerid>,<callid>,<callrecord guid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<callrecord guid> = guid of call record in ldap db

The call info has changed please update display.

NOTE AND HISTORY AVAILABLE
NOTE AND HISTROY AVAILABLE,<partnerid>,<callid>,<callrecord guid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<callrecord guid> = guid of call record in ldap db

The call has some notes and history to collect.

NOTE ADDED TO CALL
NOTE ADDED TO CALL,<partnerid>,<callid>,<guid>,<isCall>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<guid> = guid of call record or contact in ldap db
<isCall> = 1 if is a call else is a contact

A note has been added pertaining to this call.

AUTO CONFERENCE
AUTO CONFERENCE,<partnerid>,<callid>

<partnerid> = id assigned by the partner or 0

<callid> = id assigned by client or 0

This call should auto conference in with the current call.

UPDATE USER MESSAGES

UPDATE USER MESSAGES,<partnerid>,<callid>,<name>,<num msgs>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<name> = name of mailbox
<nummsgs> = a string formatted as msgs,missed,new

Update the display for user messages

HOOK STATE

HOOKSTATE,0,0,<state>

<state> = 1 (Off Hook) or 0 (On Hook)

Reflects the physical state for POTS phones only.

AUTO LISTEN

AUTO LISTEN,<partnerid>,<callid>,<fnu1>,<fnu2>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<fnu1> = field not used
<fnu2> = field not used

This call should enter a one-way auto conference (listen only) with the current call.

BLANK

BLANK,<partnerid>,<callid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

Normal call recording has been interrupted with blanking “beeps” being inserted instead.

UNBLANK

UNBLANK,<partnerid>,<callid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

Insertion of blanking “beeps” cancelled and normal call recording resumed.

Commands

LOGIN

LOGIN,<username>,<password>,<partnercode>

<username> = optional – name to log in as
<password> = required if user name given (login code)
<partnercode>= this is the code configured against the phone port

Login / Authenticate for this session. The partnercode must match the phone configuration.

LOGOUT

LOGOUT

Logout remove authentication for this session. This command is not required is disconnecting.

NEWCALL

NEWCALL,<0>,<partnerid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

Create a new call to perform tasks on. E.g. make a call

DELCALL

DELCALL,<callid>,<partnerid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

The partner has deleted a call from its records.

DIAL

DIAL,<callid>,<partnerid>,<name>,<number>,<remextn>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<name> = name of contact or empty
<number> = number to dial
<remextn> = special field for remote id (e.g. voicemail)

Dial a number – used for the initial call setup. Name and number can be blank to allow dial tone to be delivered.

DIALDIGITS

DIALDIGITS,<callid>,<partnerid>,<digits>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<digits> = digits to send

Dial more digits for a call on overlapped dialling mode.

CONNECT

CONNECT,<callid>,<partnerid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

Answer the current call. This may not work properly if the terminal in not off-hook. The PCS 570/560/410/400/100 IP Phone and PCS 60/50 IP Softphone will automatically go off-hook.

CONNECT_TONE

CONNECT_TONE,<callid>,<partnerid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

Answer the current call and play tone. This may not work properly if the terminal in not off-hook. The PCS 5xx/410/400/100 IP Phone and Navigate/PCS 60/50 IP Softphone will automatically go off-hook.
RELEASE
RELEASE,<callid>,<partnerid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

This is the clean way to hang up a call.

PARK

PARK,<callid>,<partnerid>,<name>,<parkedto>,<slot>

<partnerid> = id assigned by the partner or 0

<callid> = id assigned by client or 0
<name> = name or number of item to park (visual)
<parkedto> = parked to guid (company guid)
<slot> = number between 1 and 65535

Park this call onto the system. It will be cleared down automatically.

PICKUP

PICKUP,<callid>,<partnerid>,<name>,<partedto>,<slot>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<name> = name or number of item to park (visual)
<parkedto> = parked to guid (company guid)
<slot> = number between 1 and 65535

Pickup a parked or ringing call (name would = department or user name for ringing calls). This is executed on a real call which will be cleared and a new incoming call will appear marked for pickup. (clients will automatically answer it).

TRANSFER

TRANSFER,<callid>,<partnerid>,<source callid>,<source call partnerid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<source callid> = id of source call to transfer over
<source partnerid>

Transfer this call and the specified source call together.

WRAPUPDONE

WRAPUPDONE,<callid>,<partnerid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

Indicate that wrap has been completed early.

SENDTOVOICEMAIL

SENDTOVOICEMAIL,<callid>,<partnerid>

<partnerid> = id assigned by the partner or 0

<callid> = id assigned by client or 0

Divert this call straight to voicemail.

CURRENTCALL

CURRENTCALL,<callid>,<partnerid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

Make this call the current call – switch voice path etc.

ACCOUNTCODE

ACCOUNTCODE,<callid>,<partnerid>,<code>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<code> = set the account code for this call

Set the account code for the current call – and allow is the proceed to make the call.

RECORD

RECORD,<callid>,<partnerid>,<name>,<number>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0
<name> = name of mailbox to record to
<number> = number of source to record as

Start recording or stop recording if already in progress.

CONFERENCE

CONFERENCE,<callid>,<partnerid>

<partnerid> = id assigned by the partner or 0
<callid> = id assigned by client or 0

Begin conference or breakout of conference if already in conference.

BLANK

BLANK,<callid>,<partnerid>

<callid> = id assigned by client or 0
<partnerid> = id assigned by the partner or 0

Normal call recording will be interrupted with blanking “beeps” being inserted. instead.

UNBLANK

UNBLANK,<callid>,<partnerid>

<callid> = id assigned by client or 0
<partnerid> = id assigned by the partner or 0

Insertion of blanking “beeps” will be cancelled and normal call recording resumed.

Appendix 1

Q.931 & H.323 Cause Codes

1
Unallocated (unassigned) number
Typical scenarios include:
•The number is not in the routing table, or it has no path across the ISDN network.
Indicates that the destination requested by the calling user cannot be reached because the number is unassigned.

2
No route to specified transit network (national use)
Typical scenarios include:
•The wrong transit network code was dialed.
•The transit network does not serve this equipment.
•The transit network does not exist.
Indicates that the gateway is asked to route the call through an unrecognized intermediate network.

3
Destination address resolution failure
Typical scenarios include:
•Domain Name System (DNS) resolution failure
•Invalid session target in configuration
CC_CAUSE_NO_ROUTE
Indicates that the called party cannot be reached because the network that the call has been routed through does not serve the desired destination.

4
Send special information tone
Typical scenarios include:
•The dialed number has a special condition applied to it.
Indicates that the called party cannot be reached for reasons that are of a long-term nature and that the special information tone should be returned to the calling party.

5
Misdialed trunk prefix (national use)
Typical scenarios include:
•The wrong trunk prefix was dialed.
Indicates the erroneous inclusion of a trunk prefix in a called party number.

6
Channel unacceptable
Typical scenarios include:
•Failed channel on the network.
Indicates that the channel most recently identified is not acceptable to the sending entity for use in this call.

7
Call awarded and being delivered in an established channel
Typical scenarios include:
•Successful call.
Indicates that the user has been awarded the incoming call and that the incoming call is being connected to a channel already established to that user for similar calls.

8
Preemption
Typical scenarios include:
•Emergency services
Indicates the call is being pre-empted.

9
Preemption. Circuit reserved for reuse
Typical scenarios include:
•Emergency services
Indicates the call is being pre-empted and the circuit is reserved for reuse by pre-empting exchange.

16
Normal call clearing
Typical scenarios include:
•A call participant hung up.
Indicates that the call is being cleared because one of the users involved with the call has requested that the call be cleared.

17
User busy
Typical scenarios include:
•User is already using the telephone.
Indicates that the called party is unable to accept another call because the user busy condition has been encountered. This cause value can be generated by the called user or by the network. In the case of user determined user busy, it is noted that the user equipment is compatible with the call.

18
No user responding
Typical scenarios include:
•The user is not answering the telephone.
Used when the called party does not respond to a call establishment message with either an alerting or connect indication within the time allotted. The number that is being dialed has an active D-channel, but the far end chooses not to answer.

19
No answer from the user (user alerted)
Typical scenarios include:
•The user is not answering the telephone.

Used when the called party has been alerted but does not respond with a connect indication within the time allotted. This cause is not generated by Q.931 procedures but can be generated by internal network timers.

20
Subscriber absent
Typical scenarios include:
•The user lost network connectivity or is out of range.
Used when a mobile station has logged off, radio contact is not obtained with a mobile station, or if a personal telecommunication user is temporarily not addressable at any user-network interface.

21
Call rejected
Typical scenarios include:
•Subscriber has a service constraint that does not accept this call.
Indicates that the equipment sending this cause code does not wish to accept this call, although it could have accepted the call because the equipment sending the cause is neither busy nor incompatible.
Might also be generated by the network indicating that the call was cleared because of a supplementary service constraint. The diagnostic field might contain additional information about the supplementary service and reason for rejection.

22
Number changed
Typical scenarios include:
•A subscriber has changed their number.
Returned to a calling party when the called number indicated by the calling party is no longer assigned. The new called party number might be optionally included in this diagnostic field.

23
Redirection to a new destination
Typical scenarios include:
•Call is forwarded
Used by a general ISUP protocol mechanism that decides that the call should be sent to a different called number.

25
Exchange routing error
Typical scenarios include:
•Network is overloaded
Indicates that the destination indicated by the user cannot be reached because an intermediate exchange has released the call due to reaching a limit in executing the hop counter procedure.

26
Nonselected user clearing
Typical scenarios include:
•Called number failure

Indicates that the user has not been awarded the incoming call.

27
Socket failure
Typical scenarios include:
•Transmission Control Protocol (TCP) socket connection failure
•Problem sending an H.323 SETUP
•Problem sending a Session Initiation Protocol (SIP) INVITE
•Send or receive error occurs on connected socket
CC_CAUSE_DESTINATION_OUT_OF_ORDER
Indicates that the destination indicated by the user cannot be reached because the destination's interface is not functioning correctly.
The signaling message cannot be delivered to the remote party.

28
Invalid number format
Typical scenarios include:
•The caller is calling out using a network type number (enterprise) rather instead of Unknown or National.
Indicates that the called party cannot be reached because the called party number is not in a valid format or is not complete.

29
Facility rejected
Typical scenarios include:
•A network service is not functioning.
Indicates that a supplementary service requested by the user cannot be provided by the network.

30
Response to STATUS ENQUIRY
Typical scenarios include:
•A STATUS message is returned.
Included in the STATUS message when the reason for generating the STATUS message was the prior receipt of a STATUS ENQUIRY message.

31
Normal, unspecified
Typical scenarios include:
•Normal operation
Reports a normal event only when no other cause in the normal class applies.

34
No circuit/channel available
Typical scenarios include:
•No B-channels are available to make the selected call.
Indicates that there is no appropriate circuit or channel presently available to handle the call.

38
Network out of order
Typical scenarios include:
•Network failure.
Indicates that the network is not functioning correctly and that the condition is likely to last for an extended period.

39
Permanent frame mode connection is out of service
Typical scenarios include:
•Equipment or section failure.
Included in a STATUS message to indicate that a permanently established frame mode connection is out of service.

40
Permanent frame mode connection is operational
Typical scenarios include:
•Normal operation.
Included in a STATUS message to indicate that a permanently established frame mode connection is operational and capable of carrying user information.

41
Temporary failure
Typical scenarios include:
•Network failure.
Indicates that the network is not functioning correctly and that the condition is likely to be resolved quickly.

42
Switching equipment congestion
Typical scenarios include:
•High traffic
Indicates that the switching equipment generating this cause is experiencing high traffic.

43
Access information discarded
Typical scenarios include:
•Usually reported when the far-end ISDN switch removes some piece of information before tandem-switching a call.
Indicates that the network could not deliver access information to the remote user as requested.

44
Requested circuit/channel not available
Typical scenarios include:
•Occurs during glare condition when both sides are selected top-down or bottom-up. Change the Allocation Direction so that one end is top-down and the other is bottom-up.

Returned when the circuit or channel indicated by the requested entity cannot be provided by the other side of the interface.

46
Precedence call blocked
Typical scenarios include:
•Caller is busy and the priority level of active call is equal or higher than the incoming call.
Indicates that there are no pre-emptable circuits or that the called user is busy with a call of equal or higher pre-emptable level.

47
Internal resource allocation failure
Typical scenarios include:
•Out of memory
•Internal access to the TCP socket is unavailable
CC_CAUSE_NO_RESOURCE
Indicates a "resource unavailable" event.

49
QoS error
Typical scenarios include:
•Quality of service (QoS) error
CC_CAUSE_QOS_UNAVAILABLE
Indicates that the requested QoS cannot be provided.

50
Requested facility not subscribed
Typical scenarios include:
•The caller is trying to use a service that is not permitted.
Indicates that the user has requested a supplementary service that the user is not authorized to use.

53
Outgoing calls barred within Closed User Group (CUG)
Typical scenarios include:
•Subscriber configuration contains this limitation.
Indicates that although the calling party is a member of a CUG for the outgoing CUG call, outgoing calls are not allowed for this member of the CUG.

55
Incoming calls barred within Closed User Group (CUG)
Typical scenarios include:
•Subscriber configuration contains this limitation.
Indicates that although the called party is a member of a CUG for the incoming CUG call, incoming calls are not allowed for this member of the CUG.

57
Bearer capability not authorized

Typical scenarios include:
•The caller is not authorized to use the bearer capability.
Indicates that the user has requested a bearer capability which is implemented on the equipment but the user is not authorized to use.

58
Bearer capability not presently available
Typical scenarios include:
•A call is placed with a bearer capacity that the service provider does not have the capacity to supply.
Indicates that the user has requested a bearer capability which is implemented by the equipment and is currently unavailable.

62
Inconsistency in designated outgoing access information and subscriber class
Typical scenarios include:
•Network error.
Indicates that there is an inconsistency in the designated outgoing access information and subscriber class.

63
Service or option not available, unspecified
Typical scenarios include:
•Service not available.
Reports a service or option not available event only when no other cause in the service or option not available class applies.

65
Media negotiation failure
Typical scenarios include:
•No codec match occurred.
•H.323 or H.245 problem leading to failure in media negotiation
CC_CAUSE_BEARER_CAPABILITY_NOT_IMPLEMENTED
Indicates that the equipment sending this cause does not support the bearer capability requested.

66
Channel type not implemented
Typical scenarios include:
•Channel type match not found.
Indicates that the equipment sending this cause does not support the channel type requested.

69
Requested facility not implemented
Typical scenarios include:
•Service type match not found.
Indicates that the equipment sending this cause does not support the requested supplementary service.

70
Only restricted digital information bearer capability is available (National use)
Typical scenarios include:
•Routing error.
Indicates that the calling party has requested an unrestricted bearer service but that the equipment sending this cause only supports the restricted version of the requested bearer capacity.

79
Service or option not implemented, unspecified
Typical scenarios include:
•Service not implemented.
Reports a service or option not implemented event only when no other cause in the service or option not implemented class applies.

81
Invalid call reference value
Typical scenarios include:
•The far-end switch did not recognize the call reference for a message sent by the gateway.
Indicates that the equipment sending the cause has received a message with a call reference which is not currently in use on the user-network interface.

82
Identified channel does not exist
Typical scenarios include:
•Fractional PRI error.
Indicates a call attempt on a channel that is not configured.

83
A suspended call exists, but this call identity does not
Typical scenarios include:
•Call ID mismatch
Indicates a call resume has been attempted with a call identity which differs from that in use for any presently suspended calls.

84
Call identity in use
Typical scenarios include:
•Equipment error.
Indicates that the network has received a call suspended request containing a call identity which is already in use for a suspended call.

85
No call suspended
Typical scenarios include:
•Equipment error.
Indicates that the network has received a call resume request containing a call identity information element which does not indicate any suspended call.

86
Call having the requested call identity has been cleared
Typical scenarios include:
•Network timeout
•Call cleared by remote user.
Indicates that the network has received a call identity information element indicating a suspended call that has in the meantime been cleared wile suspended.

87
User is not a member of Closed User Group (CUG)
Typical scenarios include:
•Caller is not authorized.
Indicates that the called user for the incoming CUG call is not a member of the specified CUG.

88
Incompatible destination
Typical scenarios include:
•Number dialed is not capable of this type of call.
•Caller is calling a restricted line in unrestricted mode.
•Caller is calling a POTS phone using unrestricted mode.
Indicates that the equipment sending this cause has received a request to establish a call which has compatibility attributes which cannot be accommodated.

90
Nonexistent Closed User Group (CUG)
Typical scenarios include:
•Configuration or dialing error.
Indicates that the specified CUG does not exist.

91
Invalid transit network selection (National use)
Typical scenarios include:
•Network error.
•Identification mismatch
Indicates that a transit network identification was received which is of an incorrect format.

95
Invalid message received error
Typical scenarios include:
•An invalid message was received
CC_CAUSE_INVALID_MESSAGE
Indicates an invalid message event.

96
Mandatory IE missing error
Typical scenarios include:

•Mandatory Contact field missing in SIP message.
•Session Description Protocol (SDP) body is missing.
CC_CAUSE_MANDATORY_IE_ MISSING
Indicates that the equipment sending this cause code has received a message that is missing an information element (IE). This IE must be present in the message before the message can be processed.

97
Message type nonexistent or not implemented
Typical scenarios include:
•Message type information is missing.
Indicates that the equipment sending this cause has received a message which is missing an information element that must be present in the message before the message can be processed.

98
Message not compatible with call state or message type nonexistent or not implemented
Typical scenarios include:
•ISDN protocol mismatch
•ISDN state machine violation
Indicates that the equipment sending this cause has received a message such that the procedures do not indicate that this is a permissible message to receive while in this call state.

99
An information element or parameter does not exist or is not implemented
Typical scenarios include:
•Element mismatch
Indicates that the equipment sending this cause has received a message which includes information elements or parameters not recognized because the information element or parameter names are not defined or are defined but not implemented by the equipment.

100
Invalid IE contents error
Typical scenarios include:
•SIP Contact field is present, but format is bad
CC_CAUSE_INVALID_IE_ CONTENTS
Indicates that the equipment sending this cause code has received an IE that it has implemented. However, the equipment sending this cause code has not implemented one or more of the specific fields.

101
Message in invalid call state
Typical scenarios include:
•An unexpected message was received that is incompatible with the call state
CC_CAUSE_MESSAGE_IN_INCOMP_CALL_STATE
Indicates that a message has been received that is incompatible with the call state.

102
Call setup timeout failure
Typical scenarios include:
•No H.323 call proceeding
•No H.323 alerting or connect message received from the terminating gateway
•Invite expires timer reached maximum number of retries allowed
CC_CAUSE_RECOVERY_ON_ TIMER_EXPIRY
Indicates that a procedure has been initiated by the expiration of a timer in association with error handling procedures.

103
Parameter nonexistent or not implemented— passed on (National use)
Typical scenarios include:
•Configuration mismatch.
Indicates that the equipment sending this cause has received a message which includes parameters not recognized because the parameters are not defined or are defined but not implemented on the equipment.

110
Message with unrecognized parameter discarded
Typical scenarios include:
•Unrecognized parameter.
Indicates that the equipment sending this cause has discarded a received message which includes a parameter that is not recognized.

111
Protocol error, unspecified
Typical scenarios include:
•Protocol error.
Reports a protocol error event only when no other cause in the protocol error class applies.

127
Internal error
Typical scenarios include:
•Failed to send message to Public Switched Telephone Network (PSTN)
CC_CAUSE_INTERWORKING
Indicates that there has been interworking with a network that does not provide causes for actions it takes. Precise cause cannot be ascertained.

Issue 1.3
11th September 2019