Quality parameters and standards for Telecommunication networks (Standards and Inspection Guideline)

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1 Introduction

1.1 Background

This is the Inspection Guide for Bureau of Telecommunications and Post St. Maarten (BTPSXM) concerning Quality of Service (QoS) and Quality of Experience (QoE) parameters. The inspection Guide is further building upon the study 'Quality parameters and standards for Telecommunication networks'.

This document provides a description of a set of parameters for inspection and monitoring purposes that BTP could use as a reference for implementing QoS/QoE requirements. The regulatory approach and scope is not taken into account in this document. For implementing a QoS/QoE inspection and monitoring process it is needed to further consider the following issues:

1) It should be noted that since St. Maarten is a small market, and has limited resources , it may be very difficult to keep track of developments and amendments regarding quality parameters internationally, and to keep a consistent approach nationally;
2) The need for a process for inspection and monitoring may depend on the perception of the quality that currently is provided by the service providers active at St. Maarten. If the quality is perceived as sufficient (or more than sufficient) then it may be useful to define minimum levels for the quality parameters and to have a low profile monitoring and inspection scheme. However if the quality perceived is not sufficient and subject to significant improvement then a different and more strict approach will be needed;
3) The regulatory approach raises questions in particular since the Network and Service providers in St Maarten currently have a license or are operating under a concession. Particular requirements for service and network quality - irrespective of which they are – are not in scope of the license and/or concessions. This means the regulator would need a regulatory basis to impose such requirements;
4) As there is only limited experience with QoS/QoE regulation, introducing it, is not simply defining parameters and setting minimum values: the regulator should closely monitor the response from providers and users. It is well possible that the set described in this document needs regular updates to come to the intended results and understanding.
5) This document provides performance objectives for quality parameters. Those targets need to be validated and evaluated by BTP, and if needed amended.

2 Defining the set of quality parameters

In the next chapter a set of parameters is defined as a guide for inspection and monitoring. The parameters are divided over three sections:

1) General QoS performance
2) Network performance
3) Quality of service
4) Quality of experience

Each parameter has:

1) A name
2) A description
3) A reference to specification/recommendation
4) A performance objective

3 General QoS performance

3.1 Supply time for initial connection

Supply time for initial connection
The duration from the instant of a valid service order being received by a direct service provider to the instant a working service is made available for use.
The supply time for initial connection is based on ETSI EG 201 769 V1.1.2 (2000-10) and will be applied to all services.
The required supply time for initial connection is: • Fixed line (copper, coax, fiber optic): 95% of the connections within 60 days • Mobile: 95% of the connections within 1 day • Broadband on existing line: 95% of the connections within 5 days

3.2 Faults rate

Faults rate
A fault report is a report of disrupted or degraded service that is made by a customer and is attributable to the network
The faults rate should be calculated by dividing the number of valid fault reports observed during the data collection period by the average number of access lines or service registrations in the network under consideration during the same data collection period. This QoS indicator is based on ETSI EG 201 769 V1.1.2 (2000-10) and will be applied to all services.
The faults rate should be < 5% per year for each service.

3.3 Fault repair time

Fault repair time
A fault report is a report of disrupted or degraded service that is made by a customer and is attributable to the network
The duration from the instant a fault has been notified by the customer to the published point of contact of the service provider to the instant when the service element or service has been restored to normal working order. This measure applies only to services that offer the "standard repair" times to customers. Cases where the service provider agrees with the customer to provide faster repair for payment of higher maintenance fees are excluded, as are cases where lower fees are charged in return for a lower level of repair service. This QoS indicator is based on ETSI EG 201 769 V1.1.2 (2000-10) and will be applied to all services.
The fault repair time should be < 2 days for 80% and < 4 days for 95% of the cases.

3.4 Billing correctness complaints

Billing correctness complaints
The proportion of bills resulting in a customer complaining about the correctness of a given bill. A bill correctness complaint is an expression of dissatisfaction with a bill received from a customer. A bill correctness complaint should not be confused with a billing query (a request for information) or with a fault report.
The percentage of bills resulting in a customer complaint should be provided. Statistics should include all billing complaints received in the reporting period, regardless of the validity of the complaint and the dates of calls or any other occurrences that are the subject of the complaint. This QoS indicator is based on ETSI EG 201 769 V1.1.2 (2000-10) and will be applied to all services.
The billing correctness complaints should be < 0.3%

4 Network performance

Network Performance is specified in terms of objective performance parameters, i.e. they are measurable (with instruments or observations) and a performance value is assigned quantitatively.

4.1 Network Latency

The network latency is defined by the time it takes packets of data to make a full circuit from origination to destination and back to the originating point. This is also referred to as the round trip time (RTT). Alternatively the one-way latency could be measured (from source to destination)

Network Latency
The Network Latency is defined by the time it takes packets of data to travel a full circuit back to the originating point this is also called the round trip time (RTT) The total Network Latency can be divided in a national and an international segment. The national network latency is measured between the subscriber unit and the ISP node. The International latency is measured between the ISP node and the first network access point (in the USA.). Local ISP's should peer locally in order to reduce national latency below the international target. An example of (round-trip) Network Latency is a Ping test that measures network latency by sending an ICMP (Internet Control Message Protocol) echo request to a specified server. The shorter the Latency is the better is the performance.
It should be noted that ITU Y 1540 describes the IP Transfer Packet Delay (IPTD) across a section of a network. The Network Latency is the sum of all IPTD on a round trip. The RTT is described in ETSI EG 202 765-3 V1.1.2 and ETSI EG 202 057-4 V1.2.1
The required National Network Latency (fixed broadband), Round Trip Time, should be less than **25** (for at least 90% of the case) and the international network latency should be less than **85** ms (for at least 90% of the case). For Mobile networks the latency is depending on the radio technology applied: GPRS : < 700 ms (for at least 90% of the case) UMTS: < 200 ms (for at least 90% of the case) HSPA : < 100 ms (for at least 90% of the case) LTE: < 50 ms (for at least 90% of the case)

4.2 Throughput

Throughput
The ITU defines throughput as “an amount of user information transferred in a period of time” (ITU-T X.641). Generally ‘Throughput’ is referred to as ‘download’ or ‘upload’ speed, and it is the actual quantity of useful data transmitted during a defined time period. Throughput, or download / upload speed, changes depending on the location of the server that holds the content. The location of the content holding server may affect the throughput. In practice service providers usually provide ‘best-effort’ maximum throughput values that depend on the type of service offered to the customer.
ITU-T X.641 and ETSI EG 202 057-4 V1.2.1 (2008-07)
The actual Throughput measured (in bit per second) could be expressed as the quality parameter (preferably as a mean value over a longer period of time). This applies both the download and the upload performance. In addition the actual Throughput measured could be compared to the maximum Throughput. The Performance objective should then be expressed as a percentage of the Maximum Throughput over a longer period of time. E.g. if the (mean) Throughput is measured at 12 Megabit/sec for a service that is offered as 20 Megabit per sec, then the percentage of the Maximum Throughput is 60%. In case of a ‘guaranteed’ Throughput, this percentage could be calculated as the percentage of the guaranteed Throughput. For such service offering the performance objective percentage should be at least 100%. The test set-up and the TCP settings shall be as described in ETSI EG 202 057-4 V1.2.1 (2008-07) Annex B. For a ‘Best effort’ service, for the Throughput no performance objective is to be set.

5 Network oriented QoS Parameters:

The key network-oriented quality of service parameters are network availability, contention ratio, bandwidth utilization, and network latency.

5.1 Network Availability

Network Availability is the measure or the degree to which the access network is operable and not in a state of failure or outage at any point in time. Network Availability is a calculated value that represents the total downtime of a network, including the availability of network peripherals such as switches, multiplexers, routers, e-mail facilities (if provided) and connection to the (Internet) backbone over a month. All scheduled downtime for the purposes of maintenance and upgrading of the network system will be excluded from the calculation.

It may need a more detailed and customized description as to how the availability is calculated since most of network failure will be partial failures or local or regional failures – as will there be differences between fixed and mobile networks

Network Availability Requirement
Network Availability = [(Total Operational minutes - Total minutes of service downtime) / Total operational minutes] x 100%
ITU-T G.911
The result of this calculation should be at least 99,5% or higher. In case of mobile services this will be applied to service as such, not the individual basestation.

In IP Networks the Availability is defined (Y.1540) as follows (IP service availability parameters):

Percentage IP Service Unavailability (PIU)
The percentage of total scheduled IP service time (the percentage of T_{av} intervals) that is (are) categorized as unavailable using the IP service availability function. The minimum duration of an interval of time during which the IP service availability function is to be evaluated is T_{av}. T_{av} is provisionally defined to be five minutes. A maximum level (defined by a threshold c1) for the IPLR (Packet Loss) value is provisional for estimation of the availability as defined by the PIU The threshold c1 determines the level of packet loss above which the IP network resources are (temporarily) incapable of supporting a useful IP packet transfer service. In this case, according ITU Y.1541, values for c1 of between 0.03 and 0.2 (based on resilience of different speech coders) have been suggested for services offering Y.1541 class 0 or class 1, and c1 of 0.75 for ITU-T Y.1541 class 5

Y.1540
The result of this calculation should be 0,5 % and lower.

Percent IP Service Availability (PIA)
The percentage of total scheduled IP service time (the percentage of T_{av} intervals) that is (are) categorized as available using the IP service availability function: PIU = 100 – PIA T_{av} is provisionally defined to be five minutes.
Y.1540
The result of this calculation should be 99,5% and higher.

6 Quality of Experience

6.1 Video

The parameters for video services are:

Video Quality
There are two methods to determine the video quality: • Subjective, based on viewers rating the quality • Objective, based on actual measurements The subjective video quality depends on viewers who designate his or her opinion on a particular video sequence. Subjective video quality tests are quite expensive in terms of time (preparation and running) and human resources. ITU has described these methods in: • ITU-T J.140 (03/98), Subjective picture quality assessment for digital cable television systems • ITU-R BT.500-13 (01/2012), Methodology for the subjective assessment of the quality of television pictures • ITU-T P.913 (01/2014), Methods for the subjective assessment of video quality, audio quality and audiovisual quality of Internet video and distribution quality television in any environment ITU has also described an objective method in: The associated ITU standards are: • ITU-T J.144 (2004) “Objective perceptual video quality measurement techniques for digital cable television in the presence of a full reference.” • ITU-T J.149 (2004) “Method for specifying accuracy and cross-calibration of video quality metrics (VQM).” • ITU-R BT.1683 (2004) “Objective perceptual video quality measurement techniques for standard definition digital broadcast television in the presence of a full reference.” • ITU-T J.341 (01/2011), Objective perceptual multimedia video quality measurement of HDTV for digital cable television in the presence of a full reference Conceptually the objective method is based on using a “Full Reference” with the original quality to be compared with the actual quality after distribution.

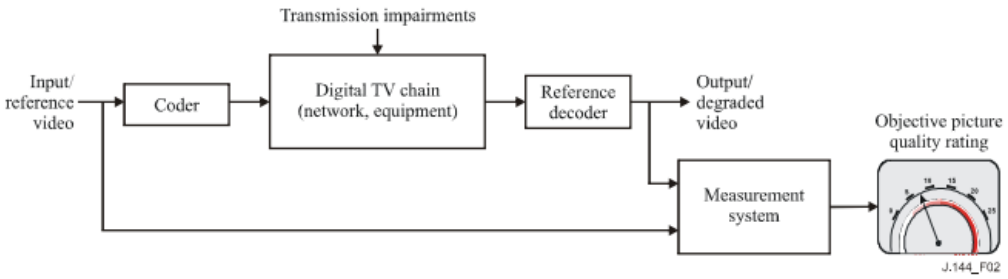


Figure 2 – Application of the full reference perceptual quality measurement method to test a transmission chain

Source: ITU-T J.144 (03/2004)

However in practice this objective method is more suitable for laboratory environments and isn't easily applied in end-user QoS/QoE measurements.

Given the lack of an implementable ITU standardized objective method at the end-user side some form of industry standard practice solution is required to implement an objective video quality measurement. ITU has specified some criteria in terms of loss events and several vendors offer a kind of video monitoring robot or electronic coach potato to simulate an end-user and measure the video quality objectively based on industry practices.

ITU-T G.1080 (12/2008) as well as TR-126, Triple-play Services Quality of Experience (QoE) Requirements, 13 December 2006 specify criteria for an acceptable number of loss events in SD and HD video. Industry practice is often using vendor specific algorithms to determine the video quality

The initial performance objective is aligned with the ITU-T G.1080 and TR-126:

- SD: Maximum one loss event per hour
- HD: Maximum one loss event per 4 hours

On top of that an overall industry practice video quality KPI will be used and calibrated for the St Maarten market.

Channel Zapping Delay
One of the key elements involved in validating quality of experience (QoE) in IPTV service is how quickly users can change TV channels, which is often referred to as channel zapping time. However, the explicit relation between channel zapping time and the user perceived quality expressed as a mean opinion score (MOS), according ITU-T G.1080 - is still under study.
The ITU-T standard G.1080," Quality of Experience Requirements for IPTV Service" and TR-126, Triple-play Services Quality of Experience (QoE) Requirements, 13 December 2006.
Time from remote control button push until stable channel is displayed on TV should be (TR-126): • < 2 [s]

The QoE Channel Zapping Delay in IPTV depends on sending of a so-called IGMP request messages to a Gateway router, and the delivery of corresponding channel data to the end point. (The time this takes is called the IGMP delay). As a Set-Top Box (STB) starts to receive IPTV multicast traffic, it stacks the packets in a buffer, which causes Buffering Delay (The time between the arrival of the first multicast traffic in the buffer and when the STB has sufficient data for playing to the screen). Once the STB starts to receive and buffer a multicast stream, there is the decoding of buffer data and rendering to the TV screen. This type of delay includes the codec decoding delay. Thus the QoE depends on a number of QoS parameters.

6.2 Internet Access

The network latency and the throughput performance are already covered in the network performance.

DNS Time
The DNS test records the time taken (in milliseconds) to resolve a fully qualified domain name to a corresponding IP address. The DNS servers used for the query are the DNS servers (primary and secondary) dynamically assigned by the service provider when the network connection is initiated. The shorter the DNS resolution time the better is the performance
Performance objective for the DNS time is not yet set, and needs to be defined in practice.

6.3 Web Browsing

The web browsing performance is a good overall indicator of the end-user experience of an Internet access service and the webpage loading time includes underlying network KPI's such as latency, packet loss, DNS performance and throughput.

Webpage Loading Time
This is the mean time to (down)load a given page from a web location. For this purpose a mix of multiple (popular) web sites will be defined. A large number of attempts are to be executed.
ETSI TS 102 250-5 V2.4.1 (2013-06) describes web browsing using HTTP
Performance objective for the Web Page Loading Time cannot be set on beforehand. First of all the value may depend on the internet access proposition. Secondly it will depend on the web sites chosen. A set of web site could be for example: www.google.com, www.youtube.com, www.facebook.com, www.cnn.com, www.twitter.com completed with the 5 most popular websites for Sint Maarten in the local CC_TLD (if relevant!)

6.4 Email

Rate of Successful Email Transfers
This is the Success rate of a given email scenario. The parameter represents the number of successful emails sent (represented as a percentage) This is the Success rate of a given email scenario. The parameter represents the number of successful emails received (represented as a percentage)
ETSI TS 102 250-5 V2.4.1 (2013-06) describes E-mail access. The QoS is also based on benchmark to other markets.
Performance objective for the Email Received Successful rate is >99% Performance objective for Email Sent Successful is >99% The size of the email is proposed to be 1 MB 1) Alternatively the QoE is differentiated between Fixed Internet access and Mobile Internet access 2) Alternatively the QoE is differentiated for different size emails e.g. 100KB, 1 MB, 5 MB This value is provisional and may need to be revised (up or down) based on real operational experience.

6.5 Voice (fixed and mobile)

Mean Opinion Score (MOS), speech quality

The quality of speech is degraded when actual voice conversation is converted to an electrical signal and transported over a network and converted back to actual voice on the other end. The used network components and the resulting network parameters will have a different impact on the measure of degradation. Speech quality measurements have mostly been subjective activities. The most commonly used method of measuring speech quality in a network is the mean opinion score (MOS) as described in the ITU-T P.800.1 recommendation.

One of the recent developments to objectively measure speech quality is POLQA described in the ITU-T P.863

ITU-T P.800.1 and ITU-T P.863 series (POLQA)

Performance objective for the MOS (POLQA NB value) on fixed networks is 3.75 for > 90%

Performance objective for the MOS (POLQA NB value) on mobile networks is 3.5 for > 90%

Whenever wideband voice services (HD Voice) are launched new thresholds will be defined for the wideband voice services using the POLQA wideband method. Once the POLQA wideband measurements becomes the default measurement method the thresholds for the narrow band voice will have to be adjusted according. The theoretical maximum MOS value is:

	POLQA NB	POLQA WB
G.711 (standard PSTN)	4.3	3.5
AMR-FR (Mobile)	4.2	3.2

Source: POLQA Technical Whitepaper, October 2011

Call Set-up Time
Call Setup Time is the mean time that is needed to set up a call to a defined destination. The call set up time is the period starting when the address information required for setting up a call is received by the network (e.g. recognized on the calling user's access line) and finishing when the called party busy tone or ringing tone or answer signal is received by the calling party (e.g. recognized on the calling user's access line).
ETSI EG 201 769 V1.1.2 (2000-10)
Performance objective for the Call Set-up Time is: • Fixed line: < 5 [s] for 95% of the cases • International: < 10 [s] for 95% of the cases • Mobile: < 8 [s] for 95% of the cases The threshold for the mobile call set-up includes the anticipated delay in call set-up caused by CSFB (Circuit Switched Call Back) at the launch of LTE. These values are provisional and may need to be revised (up or down) based on real operational experience.

Unsuccessful call ratio
Unsuccessful call ratio is defined as the ratio of unsuccessful calls to the total number of call attempts in a specified time period. An unsuccessful call is a call attempt to a valid number, properly dialled following dial tone, where neither called party busy tone, nor ringing tone, nor answer signal, is recognized on the access line of the calling user within 30 seconds from the instant when the address information required for setting up a call is received by the network. This includes many reasons for an unsuccessful call including aspects such blocking/congestion.
ETSI EG 201 769 V1.1.2 (2000-10)
Performance objective for the unsuccessful call ratio is: < 2% (all networks) and should be measured during representative periods during the day and on different days of the week/month

6.6 Mobile

Coverage PLMN (land area)
PLMN (land area) coverage is defined as “the proportion of total mobile cellular coverage of the land area in percent. This is calculated by dividing the land area covered by a mobile cellular signal by the total land area” (ITU).
GSM: Covered percentage of total land: ≥ 90% with BCCH signal strength (RSSI) of ≥ -95 dBm 3G/HSPA: Covered percentage of total land: ≥ 90% with pilot channel signal (RSCP) ≥ -105 dBm LTE: Covered percentage of total land: ≥ 90% with pilot channel signal (RSRP) ≥ -115 dBm The LTE threshold might require fine tuning depending on the mode (TDD ⇔ FDD and bandwidth 1.4 MHz up to 20 MHz) Note: The coverage requirement applies to the network, not to each individual frequency layer (for example if one frequency is used for coverage and another higher frequency for capacity then the coverage requirement only applies to the coverage layer)

Coverage PLMN (population)
PLMN (population) coverage is defined as “the percentage of inhabitants that are within range of a mobile cellular signal, irrespective of whether or not they are subscribers. This is calculated by dividing the number of inhabitants within range of a mobile cellular signal by the total population” (ITU).
All technologies require 20 dB more signal than the threshold defined in the Coverage PLMN (land area) to cater for indoor penetration in the built-up areas

Dropped Call Rate
The Dropped Call Rate is one of the most important QoS parameters for monitoring performance of a mobile network. Call attempts that successfully gained access to the network and the call setup was successful but are disconnected due to abnormal call release are called dropped calls. Dropped calls are caused by network related issues such as interference, congestion, errors in radio and signaling protocols, power attenuation, and many more. Irregular user behavior is also a cause of dropped calls and these could be subscriber unit malfunction, insufficient funds, and others. Call Set-up on first attempt and held for e.g. 2 min without dropping of the call. Rate is based on the total sample of calls that was dropped before completion.

ETSI EG 202 057-3
The Dropped Call Rate should be ≤ 2% (while in the coverage area) and should be measured during representative periods during the day and on different days of the week/month

6.7 SMS

Completion rate for SMS
Rate of Received SMS within 2 min. SMS not refused when sent out and received within 2 minutes without being altered. The Rate is to be based on the total number of SMS send attempts.
ETSI EG 202 057-2 V1.3.2 Further the QoS threshold is based on benchmarks to other markets (in Particular Kingdom of Bahrain).
Rate of Received SMS within 2 min > 95% This value is provisional and may need to be revised (up or down) based on real operational experience.

6.8 Mobile Broadband

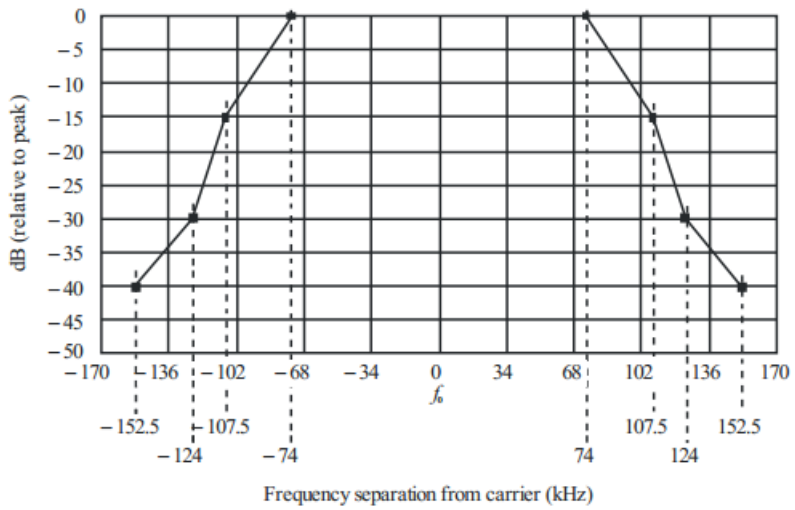
For Mobile Broadband the general throughput performance will be required. However since the difference between the peak throughput and the “typical” throughput tends to be very large, sometimes up to a factor 10, it is important that mobile operators communicate very clearly what “typical” throughput the customers can expect. The peak speed can be mentioned but should not be shown more prevalent than the “typical” speed in promotion material.

7 Broadcasting

Broadcasting services are planned for fixed, portable outdoor / indoor or mobile (e.g. car) reception and could be of different quality of service (QoS) levels. The reception can be different in different parts of a country based on geographic and topographic conditions like dense urban places, rural areas, mountainous areas and the presence of large water spaces. Broadcasting services are defined as terrestrial radio broadcasting being AM and FM-broadcasting and terrestrial and cable based TV-broadcasting.

7.1 FM Broadcasting

FM Broadcasting Coverage Specs
The following are the requirements that FM broadcasting infrastructures need to comply with in order to ensure the expected QoS and QoE for local broadcasting services
ITU-R BS.412-9 and BETS-6
• Coverage of the national territory of ≥ 90% • Coverage of the population of ≥ 90% • Signal strength of ≥ 66 dBµV/m at 10 m above ground level (ITU-R BS.412-9 urban) • Signal of carrier within ± 1000 Hz of designated carrier frequency (following the stricter BETS-6 instead of the older ITU-R SM.1045-1) • Stereophonic 38 kHz DSBSC with 19 kHz pilot • Audio bandwidth 50 Hz – 15 kHz • RDS (optional) 57 kHz carrier (CENELEC 50067) • Frequency deviation of ±75 kHz

FM Broadcasting Spurious Emissions
The following are requirements to prevent unwanted emissions in third party infrastructures
ITU-R SM.329-12, ITU-R SM.1268-3 and ITU-R SM.1541-5
• The spectrum mask should meet the ITU-R SM.1268 criteria: **Rec. ITU-R SM.1268-3** FIGURE 1 **Shape of the mask**    • More than 152.5 kHz and up to and including 500 kHz from the carrier frequency, maximum value <-40 dBc> until ITU-R SM.1541 has a final specification for this transition • More than 500 kHz from the carrier frequency < -(46 + 10 log P)> or <-70 dBc>, whichever is less stringent; the absolute mean power level of 1 mW should not be exceeded (ITU-R SM.329-12 Category A)

7.2 AM Broadcasting

AM Broadcasting Coverage Specs
The following are the requirements that AM broadcasting infrastructures need to comply with in order to ensure the expected QoS and QoE for local broadcasting services
ITU-R BS.703, BETS-5 and ITU-R SM.1045-1,
• Coverage of the national territory of ≥ 90% (mind the correction for ground conductivity) • Coverage of the population of ≥ 90% • Signal strength of ≥ 66 dBµV/m at 10 m above ground level (More if interference can be expected). For DRM a lower threshold can be applied. • Modulation depth/index ≤ 90 % • Dynamic amplitude modulation allowed • Audio bandwidth 40 Hz – 5 kHz • Signal of carrier within ± 10 Hz of designated carrier frequency (following both BETS-5 and ITU-R SM.1045-1) • Channel bandwidth ≤ 10 kHz

AM Broadcasting Spurious Emissions
The following are requirements to prevent unwanted emissions in third party infrastructures
ITU-R SM.329-12 and NRSC-2-B Emission Limitation for Analog AM Broadcast Transmission, September, 2012

- The spectrum mask should meet the NRSC-2-B criteria:

Table 1: RF Mask for Analog AM Broadcast Station Spectrum Occupancy

Frequency band relative to carrier (± kHz)	Attenuation relative to carrier (dB)
0 to 10.2	0
10.2 to 20	at least 25[3]
20 to 30	at least 35
30 to 60	–at least (5 + 1 dB/kHz) from carrier[4]
60 to 75	at least 65[4]
Above 75	at least 80[4]

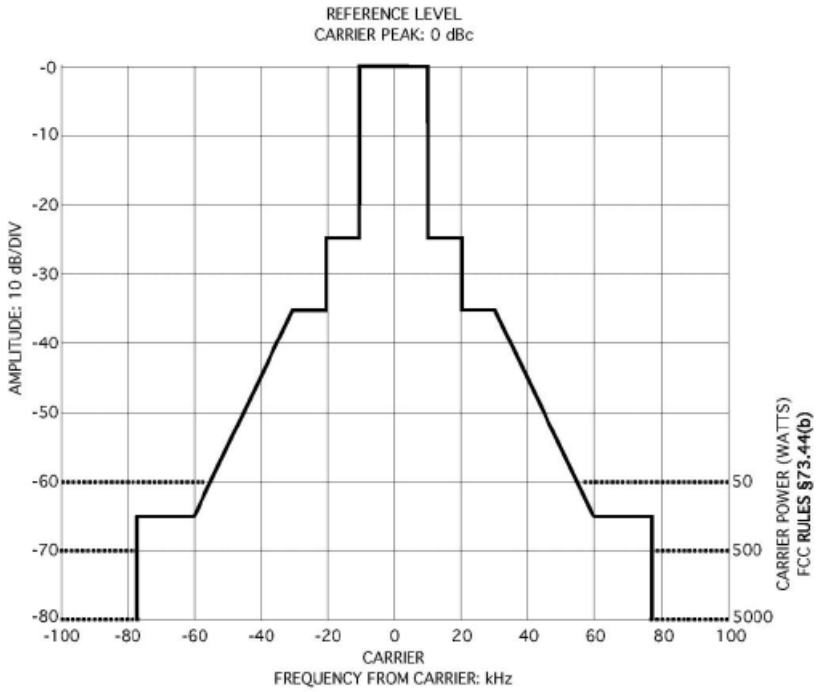


Figure 1. Analog AM Broadcast RF Emission Limits

- Beyond the spectrum mask a suppression of 43 + 10 log P or 80 dBc, whichever is less stringent, is required (ITU-R SM.329-12 Category C)

7.3 Terrestrial TV

The standard for Digital Terrestrial Television (DTT) in St. Maarten is Digital Video Broadcasting Terrestrial (DVB-T). The standard for Cable Television in St. Maarten is Digital Video Broadcasting Cable (DVB-C/C2).

DVB-T/T2 Requirements
ETSI TR 101 190 V1.3.2, ETSI TS 102 831 V1.2.1 and ISO/IEC 13818-1 -2 -3 and ETR 154

• Coverage of the national territory of ≥ 90%
• Coverage of the population of ≥ 90%
• Signal strength of ≥ 56 dBµV/m 3 m above ground level as reference for the following instances
• DVB-T for 8K mode, 64 QAM, Code Rate ¾, Guard Interval ¼
• DVB-T2 for 32K mode, 256 QAM, Code Rate $^{2}/_{5}$, Guard Interval $^{1}/_{16}$
• Single Frequency Network (SFN)
• Transmitter design specification ETSI TR 101 190
• Video:
  ○ MPEG 2 MP@ML(main profile at main level) I, P and B frames, (MPEG 4 or HEVC for DVB-T2)
  ○ Resolution 720 x 480 source NTSC minimal, 720 x 576 source PAL minimal
  ○ Minimal average video transport stream 2 Mbit/s for MPEG 2 (VBR preferred)
  ○ Audio MPEG 1, layer 2, 160 kb/s minimum
• EPG insertion required (ISO/IEC 13818-1 -2 -3 and ETR 154)
• Hierarchical modulation not allowed without specific approval of BTP

7.4 Cable TV

DVB-C/C2 Requirements

• Modulation
  ○ DVB-C min. 64-QAM
  ○ DVB-C2 min 1024-QAM
• Video:
  ○ SDTV based on PAL standard 720 x 576 min
  ○ SDTV based on NTSC standard 720 x 480 min
  ○ HDTV 1280 x 720 min
• EPG insertion required (ISO/IEC 13818-1 -2 -3 and ETR 154)
• Compression: MPEG-4/H264 or HEVC/H265
• Signal levels at any outlet 50 dBµV - 74 dBµV
• Signal quality MER ≥27 dB
BER (Pre FEC) $\leq 1 \times 10^{-6}$
SNR ≥30 dB

ANNEX A

ITU-T REC. Y.1541 provides the following guidance for the definition of network classes as follows:

Table 2/Y.1541 – Guidance for IP QoS classes

QoS class	Applications (examples)	Node mechanisms	Network techniques
0	Real-time, jitter sensitive, high interaction (VoIP, VTC)	Separate queue with preferential servicing, traffic grooming	Constrained routing and distance
1	Real-time, jitter sensitive, interactive (VoIP, VTC).		Less constrained routing and distances
2	Transaction data, highly interactive (Signalling)	Separate queue, drop priority	Constrained routing and distance
3	Transaction data, interactive		Less constrained routing and distances
4	Low loss only (short transactions, bulk data, video streaming)	Long queue, drop priority	Any route/path
5	Traditional applications of default IP networks	Separate queue (lowest priority)	Any route/path
NOTE – Any example application listed in Table 2 could also be used in Class 5 with unspecified performance objectives, as long as the users are willing to accept the level of performance prevalent during their session.			

(Snapshot from ITU-T REC.1541)